Introduction to VoIP & Asterisk

Julien Goodwin

Strategic Data

Outline

- Introduction to Telephony
- Introduction to VoIP
- – Morning Tea –
- Building a SoHo PBX with Asterisk

This talk does not cover large scale voip

Sorry about the logo's, couldn't find vector versions

An Intro to Traditional Telephony

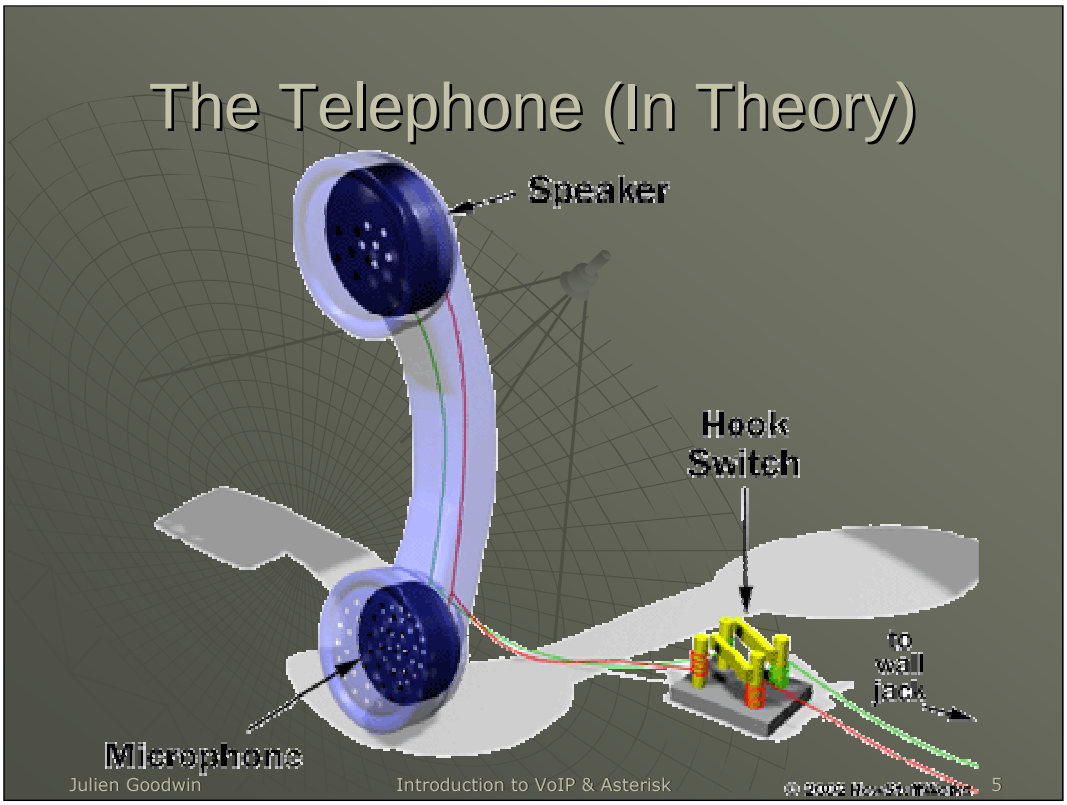
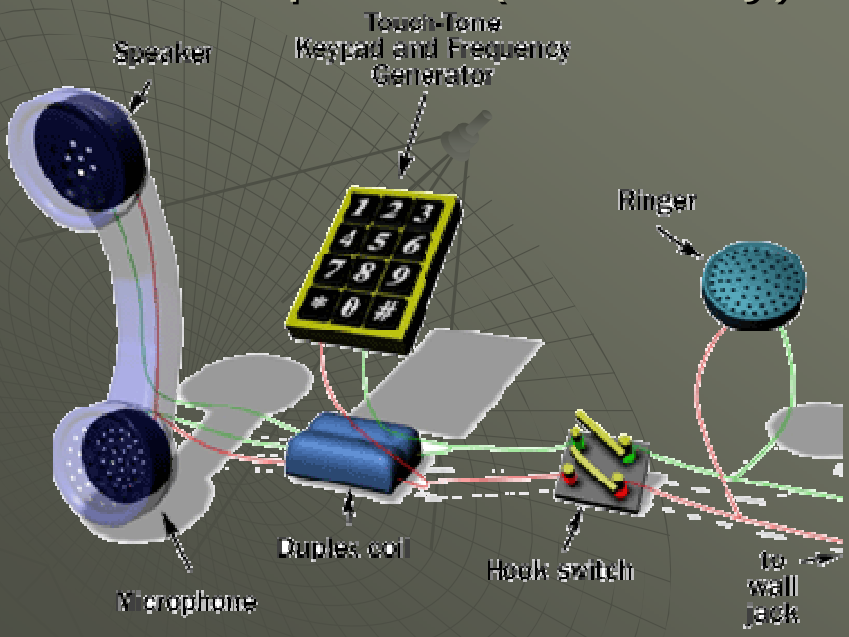
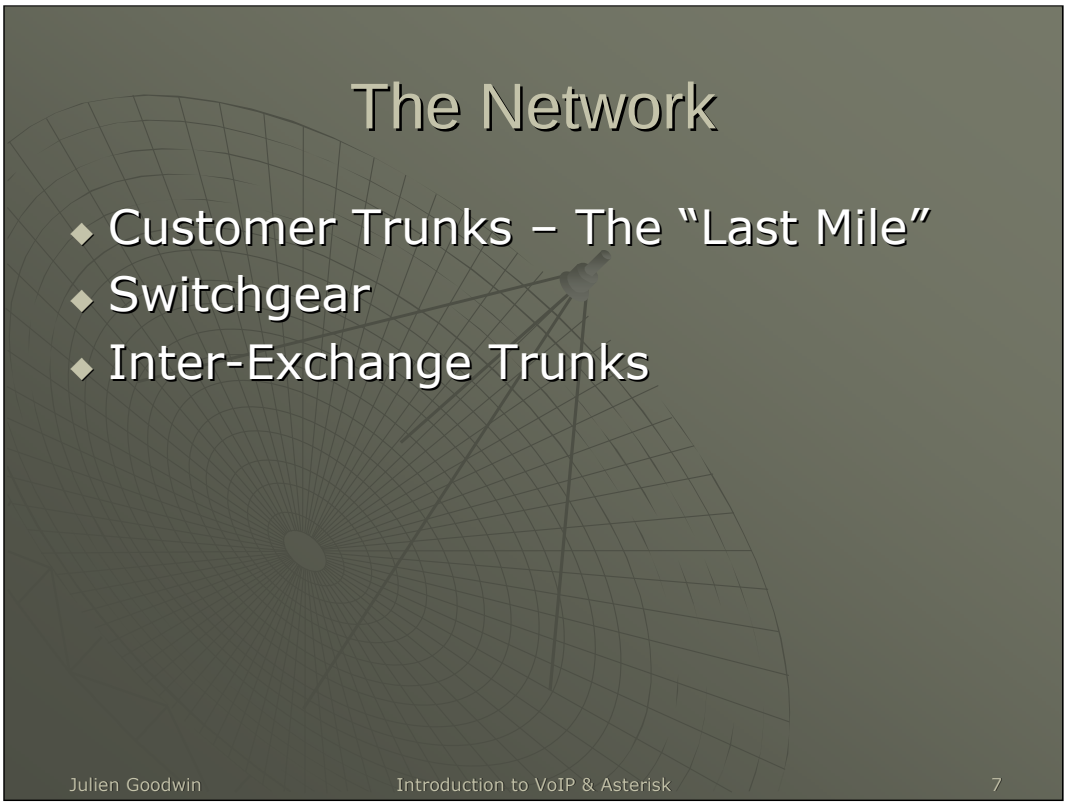


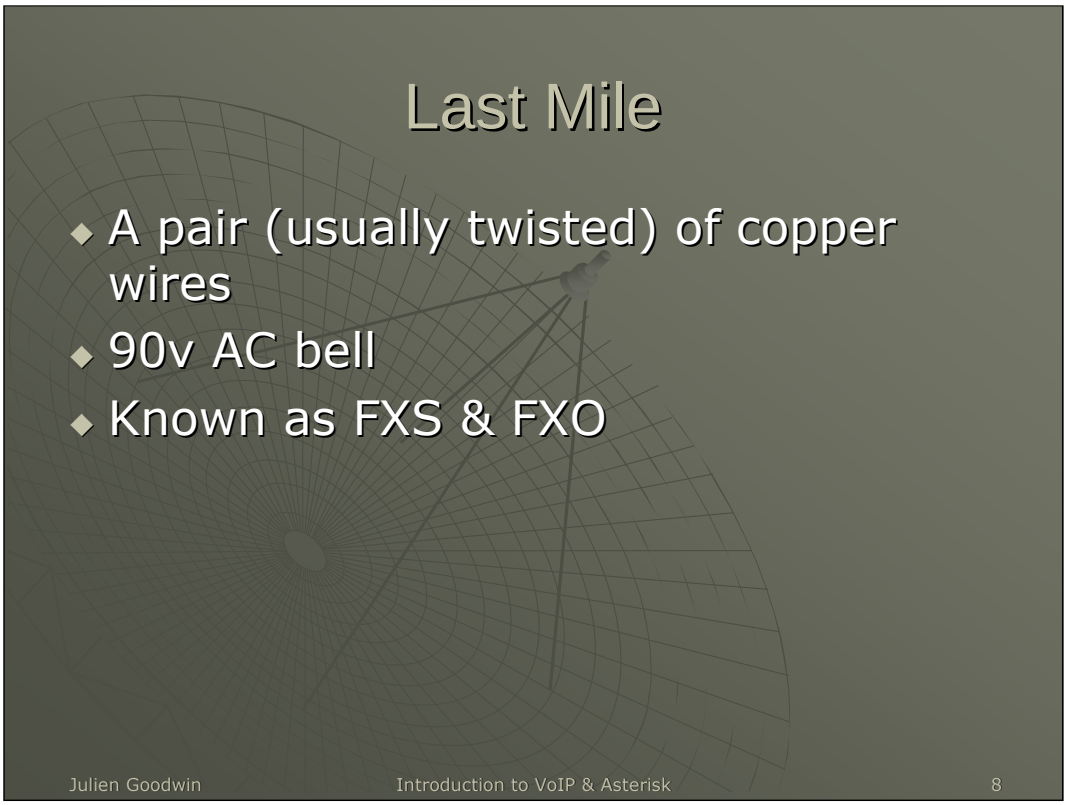
Image from howstuffworks

The Telephone (In Reality)





Switchgear – the exchange



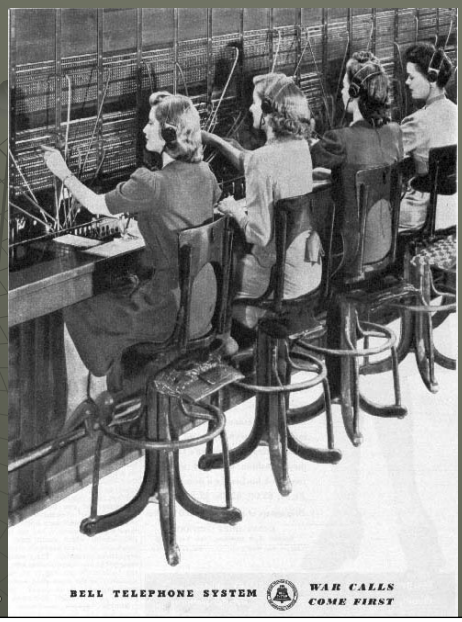
Twisted pair is actually not as important as you'd think, even DSL requires only a very slight twist. The twist is mainly to prevent crosstalk in multipair configurations

FXS – Plug a phone in to this

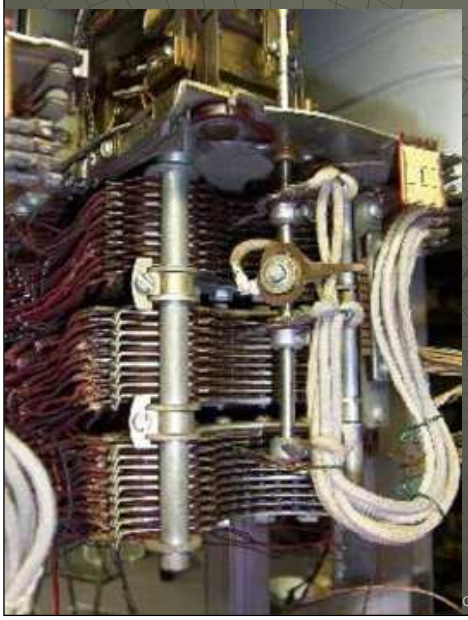
FXO – Plug a phone line in to this

Manual Exchanges

- Operator making physical connections
- Not scalable



Strowger



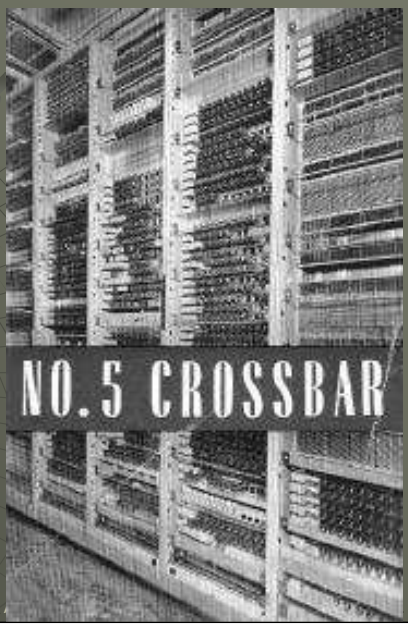
- Early Automated Exchange
- 2d selector switch
- Up to 200 contacts
- This is what pulse dialing controls

200 contacts per switch

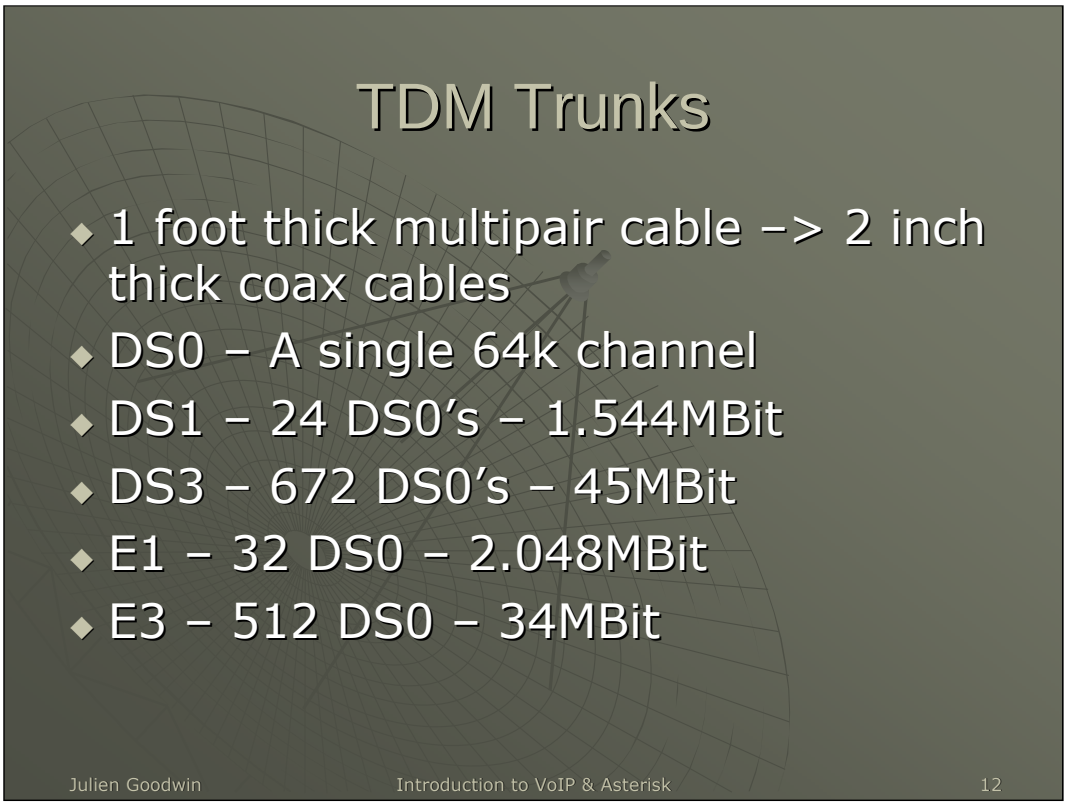
Inefficient, moving parts, large physical size

Crossbar Switching

- In on X
- Out on Y
- Electromagnets link rows
- Fewer moving parts



Higher densities, more reliable


TDM Trunks
1 foot thick multipair cable –> 2 inch thick coax cables
DS0 – A single 64k channel
DS1 – 24 DS0's – 1.544MBit
DS3 – 672 DS0's – 45MBit
E1 – 32 DS0 – 2.048MBit
E3 – 512 DS0 – 34MBit
Julien Goodwin
Introduction to VoIP & Asterisk
12

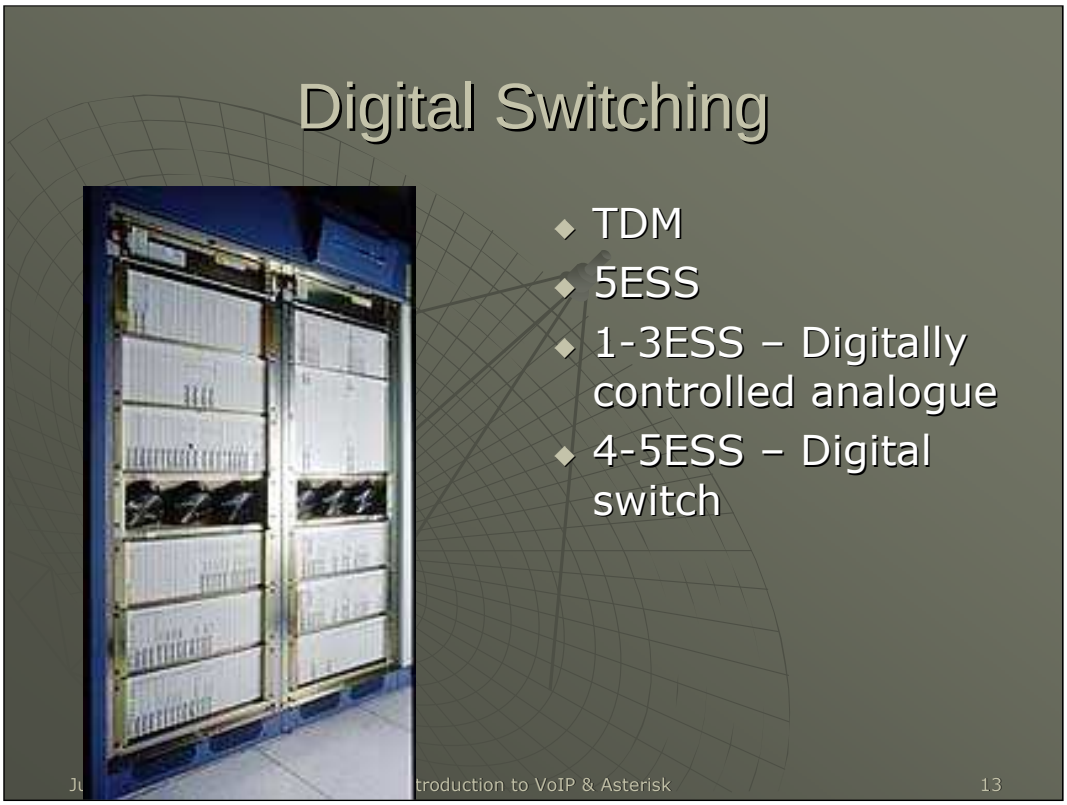
AT&T in 1957 and implemented in the early 1960's to support long-haul pulse-code modulation (PCM) voice transmission.

T-2, operating at 6.312 Mbps, (96 =4 T1)

T-4, operating at 274.176 Mbps (4032 ~ 6 T3's)

DS0 – 8000 8 bit samples

Originally, T-1 meant "Transmission - Level 1", and had to do with the media that the signal was passed over. DS-1 meant "Digital Service - Level 1", and had to do with the service to be sent(originally 24 digitized voice channels over the T1). The terms T1 and DS1 have become synonymous and include a plethora of different services from voice to data to clear channel pipes. The line speed of them is always consistent at 1.544 Mbit/s, but the payload can vary greatly.



Picture is a lucent 5ESS – in production they've hit 6 nines for six straight years

AT&T handles about 300 million voice telephone calls a day. It carried 431 million calls on Tuesday as customers flooded the telephone lines in the wake of the attacks on the World Trade Center and Pentagon, making it the heaviest business day in the company's network history, Johnson said.

AT&T's local network switching equipment, which routes telephone calls, was located in the basement of the World Trade Center towers and survived the implosion of the buildings, Johnson said.

``It appears the equipment has survived ... It was up and alive and still providing dial tone by 4 o'clock yesterday afternoon. Once the back-up batteries ran out, we took them offline, but the equipment is still working,'' Johnson said.

``We were amazed,'' he said. ``It was several stories under ground and all I can say is that they must have built up that basement very sturdy.''

The switching equipment handled calls for AT&T business customers in Lower Manhattan. The company rerouted calls and suffered no network outages, Johnson said.

AT&T will retrieve the equipment once it gets approval from New York City and disaster teams to approach the rubble of the World Trade Center. The New York-based company said none of its employees were injured or killed in the attacks.

Australia's Current Exchanges

Alcatel S12

Ericsson AXE 10

These are of course much smaller then the real exchanges

Telstra are probably migrating off the AXE to the Alcatel

ISDN

- ISDN BRI:
  - 2 DS0
  - 1 Control Channel @ 12kbps
- ISDN PRI:
  - ISDN Protocol over T/E1
  - Takes 1 DS0 for control channel

SONET/SDH

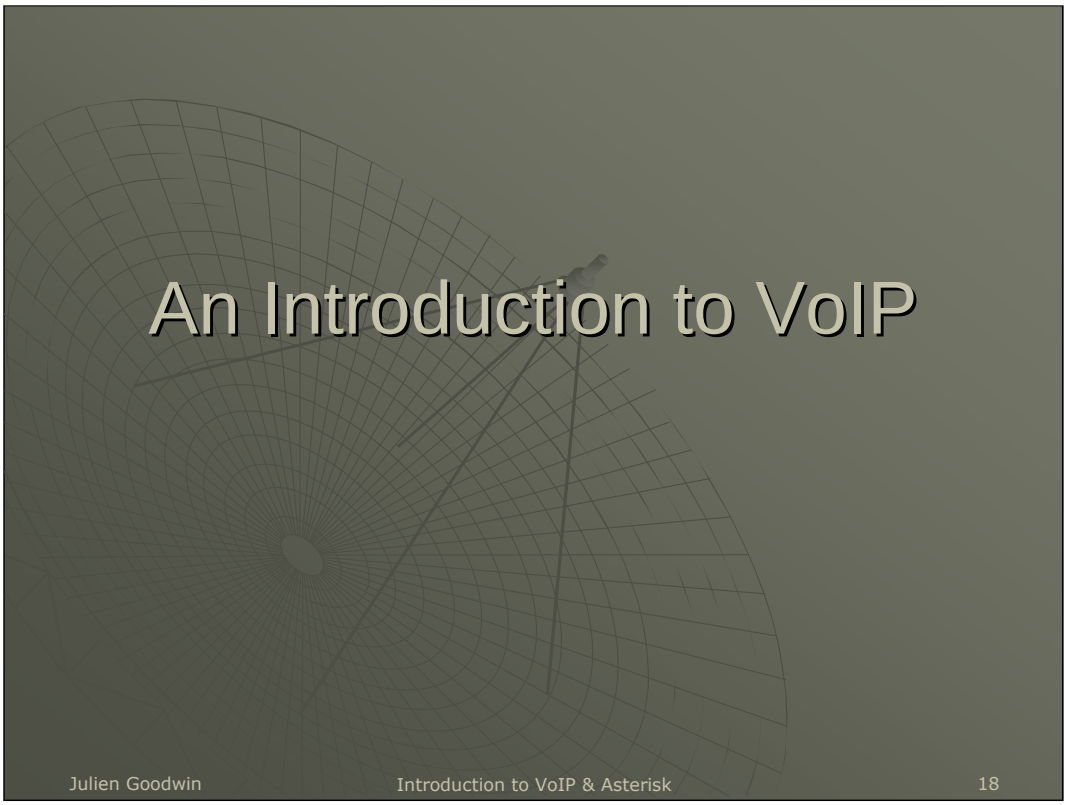
- Synchronous Optical NETwork
- Synchronous Digital Hierarchy

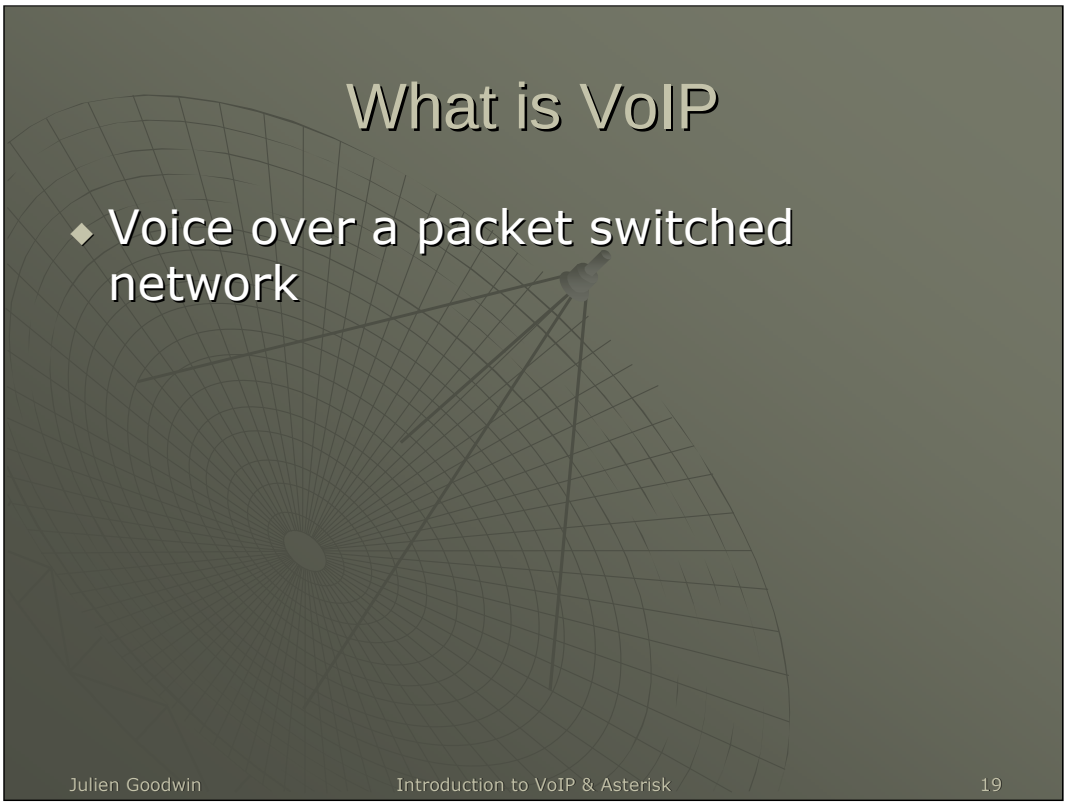
- Ring based fibre network

- OC1 - 51Mbit
- OC3 - 155Mbit
- OC12 - 622Mbit
- OC48 - 2.48Gbit
- OC192 - 10Gbit
- OC768 - 40Gbit

SS7

- 2600Hz – That was CCITT 5
- Switching control protocol


An Introduction to VoIP



Note packet switched, whilst voip technically requires IP, it can be used to encompass VoFR, VoATM and similar technologies

Today we're only dealing with genuine VoIP

VoIP Features

Toll Bypass

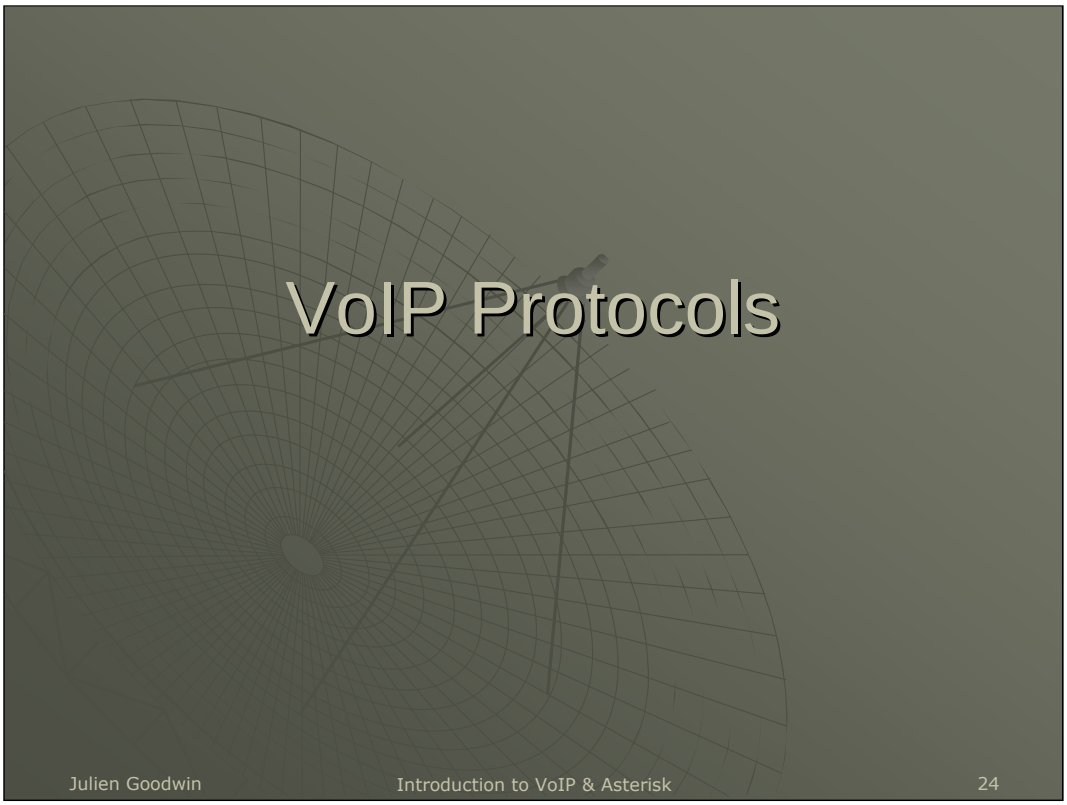
- Least cost routing
  - Eg, route mobile calls out via a mobile, not wired links
- IP Trunks
  - Between Offices
  - To Teleworkers

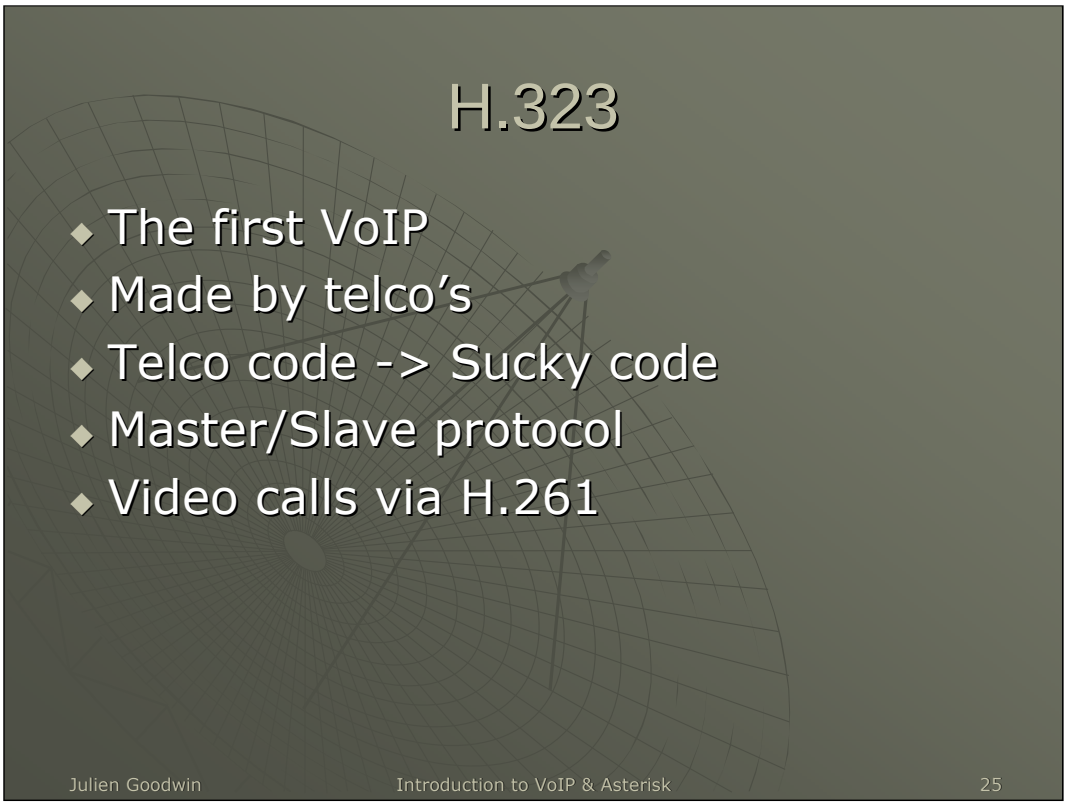
Fax over IP

- No, don't
- Standard is T.38
- Nothing implements it

VoIP PBX Features

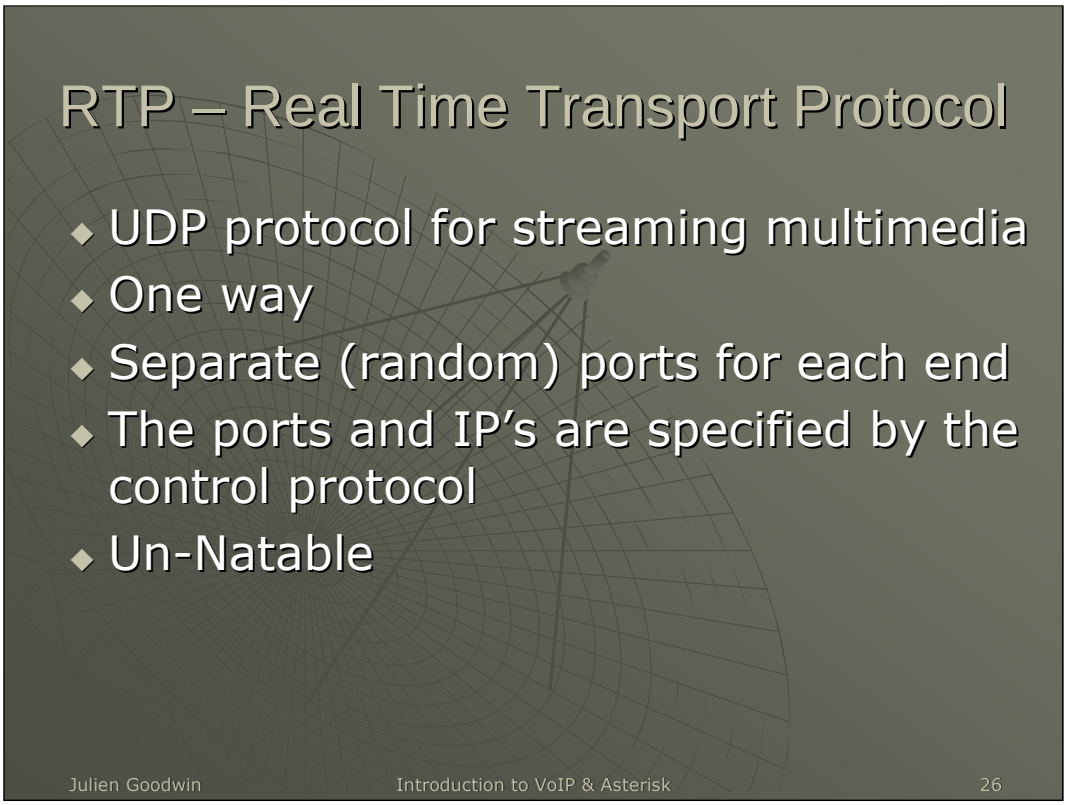
- Remote Extensions
- IP Trunks
- Extensive call routing support


VoIP Protocols



Master = gatekeeper

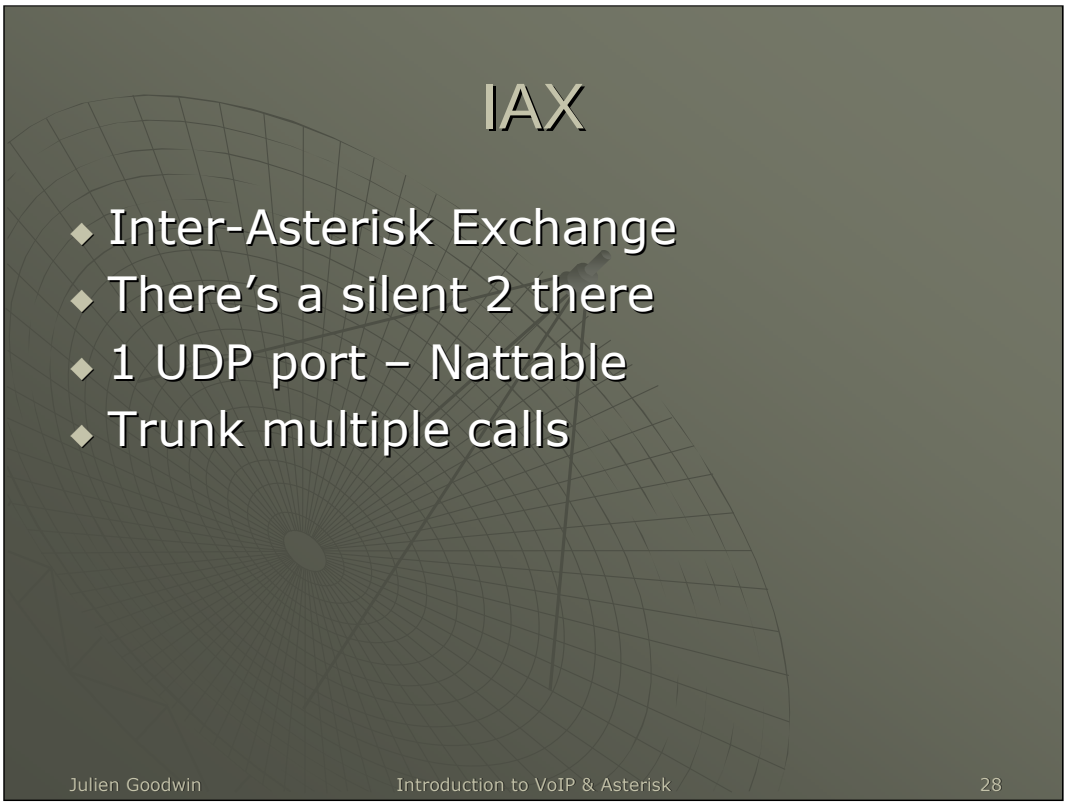
Slave = endpoint


RTP – Real Time Transport Protocol
UDP protocol for streaming multimedia
One way
Separate (random) ports for each end
The ports and IP's are specified by the control protocol
Un-Natable
Julien Goodwin
Introduction to VoIP & Asterisk
26

To nat RTP requires that the firewall support both RTP and the control protocol

SIP

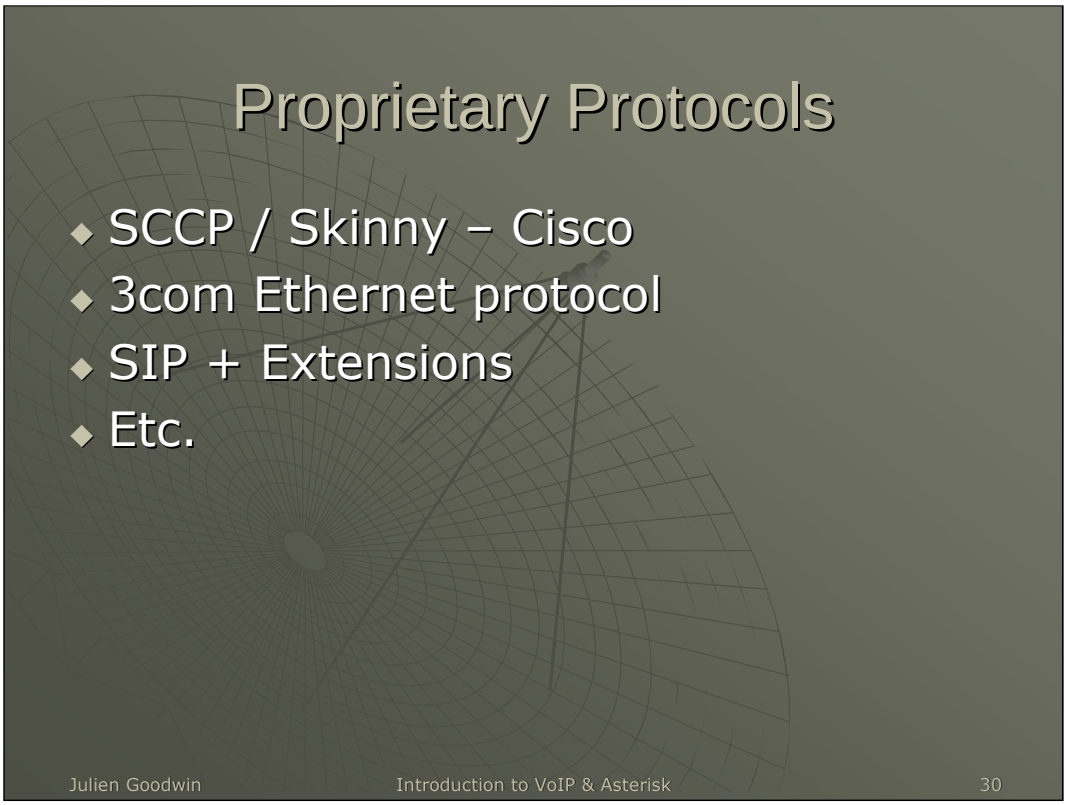
- Still uses RTP for audio/video
- HTTP-like protocol
- UDP or TCP based
- The New Standard ™



As the name implies it was created by the asterisk guys.

MGCP

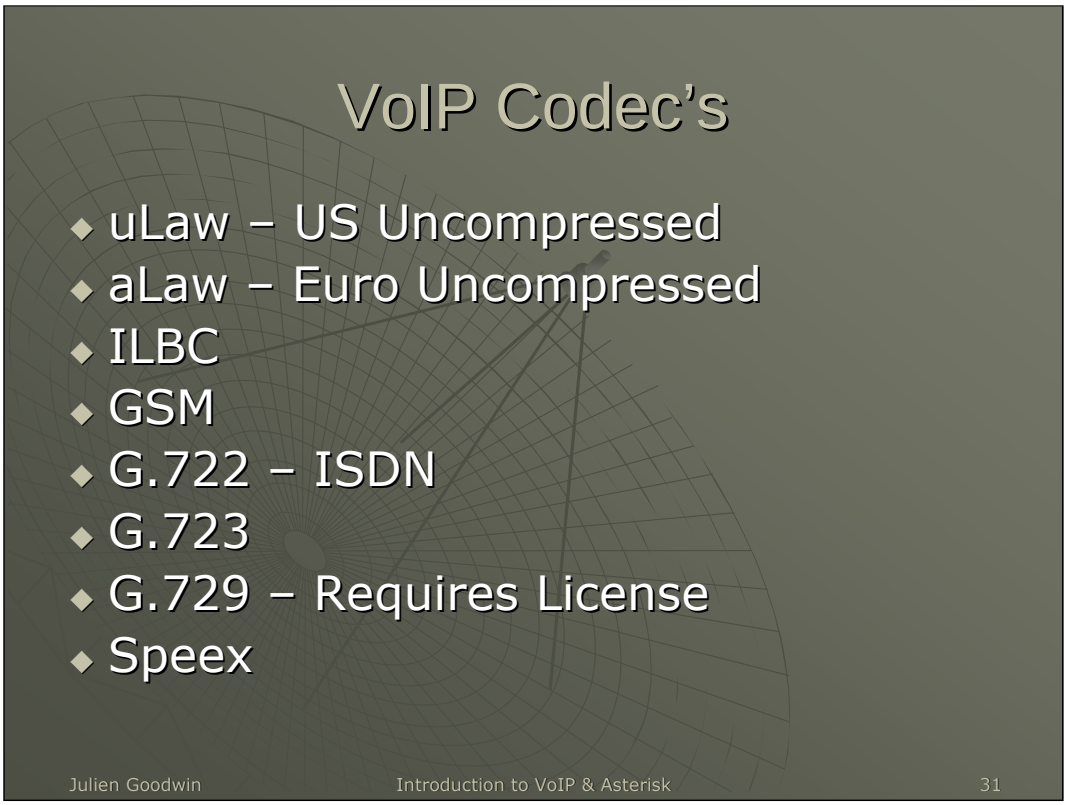
- Media Gateway Control Protocol
- Master/Slave
- Requires the PBX to provide features



3com – ETHERNET protocol, not IP, you have to pay extra for that

Be very careful with sip extensions, they're very rarely compatible between vendors

A lot of other vendors still have their own proprietery protocols that are not going away, eg avaya



These codecs are all supported by asterisk in some form

Ulaw & alaw are both 64k streams – 8000 8bit samples

Internet Low Bandwidth Codec – lower bitrates only, higher ones are patented

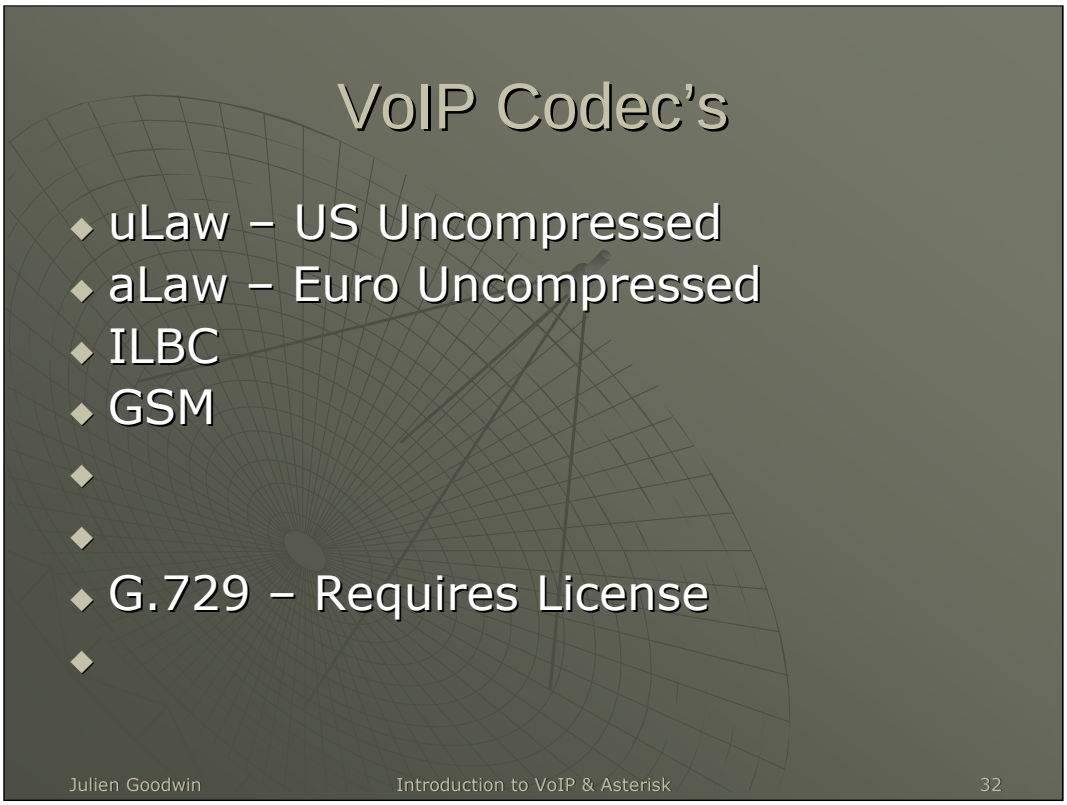
GSM – the mobile codec, the asterisk version uses only the older version which is out of patents,

722 – 48, 56, 64

723 – 5.3 & 6.4

729 – 8, requires US$10 per channel

Speex – variable, only 6mb/hour



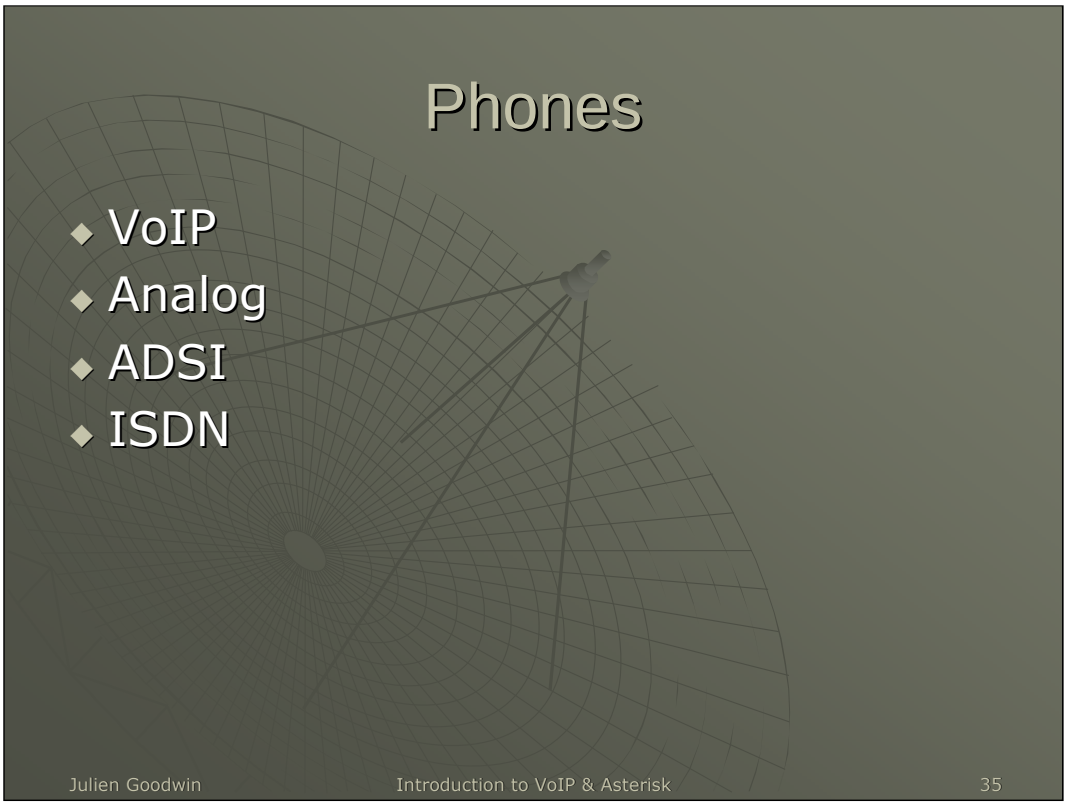
These are the ones people should care about

A short word on QoS

- It doesn't really work
- By the time it's needed you're already screwed
- Can be effective on smaller links when running on TX of both ends

Components of a VoIP PBX

- Phones
- PBX Software
- PSTN Integration



Voice over ip, which usually means ethernet

ADSI, which you can think of as analog with a display

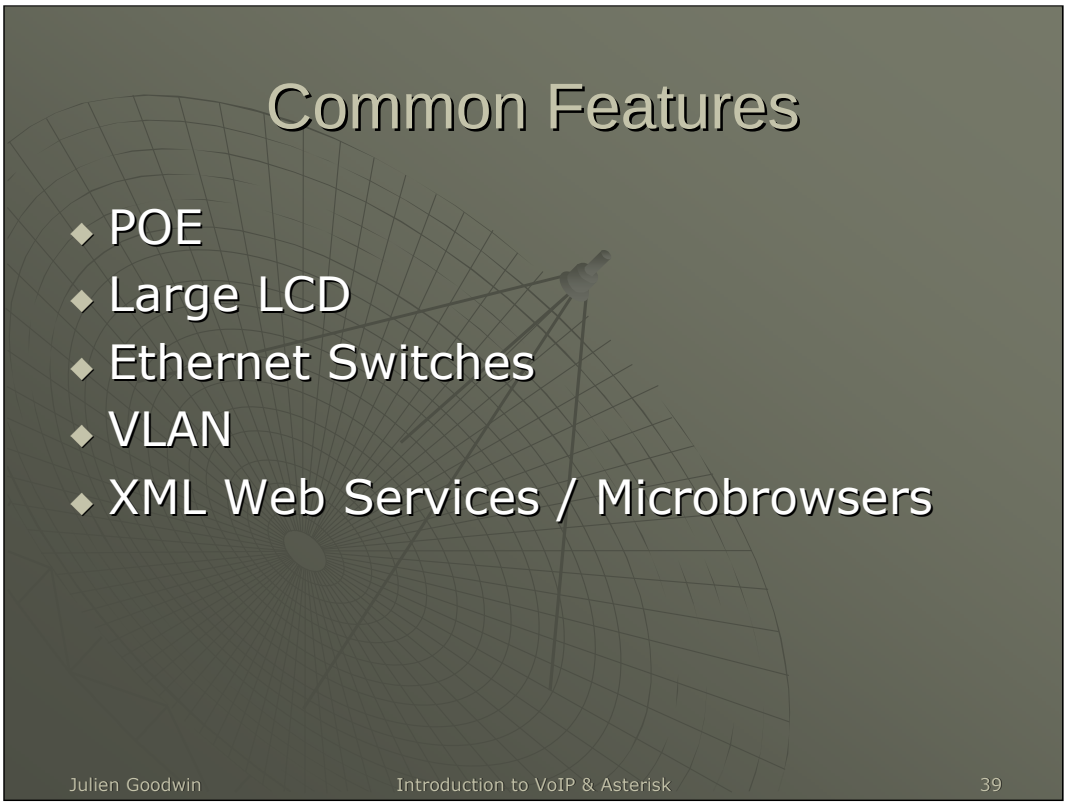
PBX Software

- Free or Commercial
- Tied to hardware or not

PSTN Integration

- Analog Lines – FXO
- ISDN – BRI/PRI
- TDM – T/E 1/3
- IP Trunks
  - H.323
  - SIP
  - IAX

VoIP Phones



Vlan support allows voice and data to go on different vlans, which can be used to have separate uplinks for voip preventing upstream contention

From the fat left:

Cisco 7902 – the one without the display

Cisco 7905 – below the 7902, lcd display and listen only speakerphone

Cisco 7912 – A 7905 with an ethernet switch

Cisco 7920 – wireless ip phone

Cisco 7940 – the two button large screen phone

Cisco 7960 – six button phone

Cisco 7914 – the extra button addon

Cisco 7970 – the colour screen phone, which is AU$1300 retail

MISSING - 7980 video phone

MISSING - 7935/6 mushroom conference phone

Note that the speakerphones are all OEM’d from polycom

These are the ones that support sip, the reast only support SCCP, and while there is an SCCP module for * it’s not quite as solid as SIP

NOTE: This is now out of date, Cisco are finally starting to release SIP images for all phones.

On the right is the 941, notice the design influence?

The one on the bottom is the linksysone phone, SIP based, not yet running with asterisk

Polycom best manufacturer

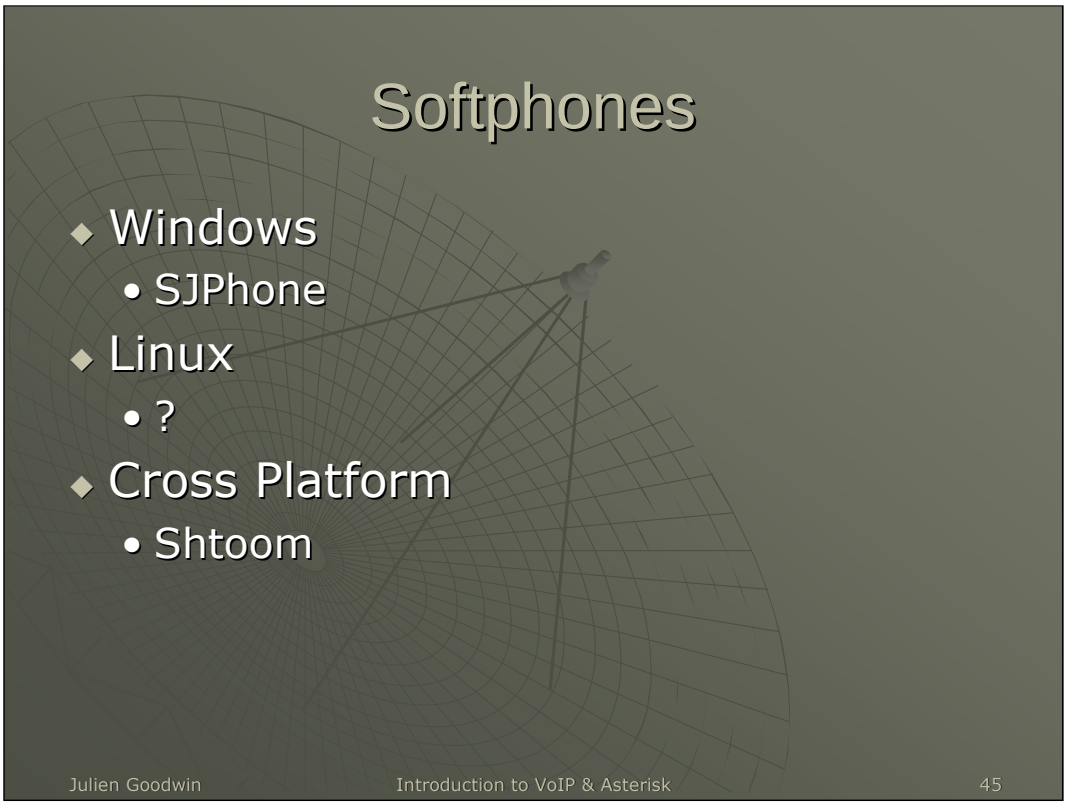
NOTE: Images appear in clockwise order, 300/500/600/600 button bank/4000 AKA cisco 7936

Mentioned the cisco 7936 before, well the ip 4000 is the unit that cisco rebadge

All these phones are native SIP, only the ip 600 has internal POE support, the rest require a custom cable

Don’t by the BT-101, it sucks. The new revision using the buttons from a GXP-2000 is physically nicer, but still not great

Grandstream firmware historically dodgy, people often complain about units bricking after firmware updates



Theory:

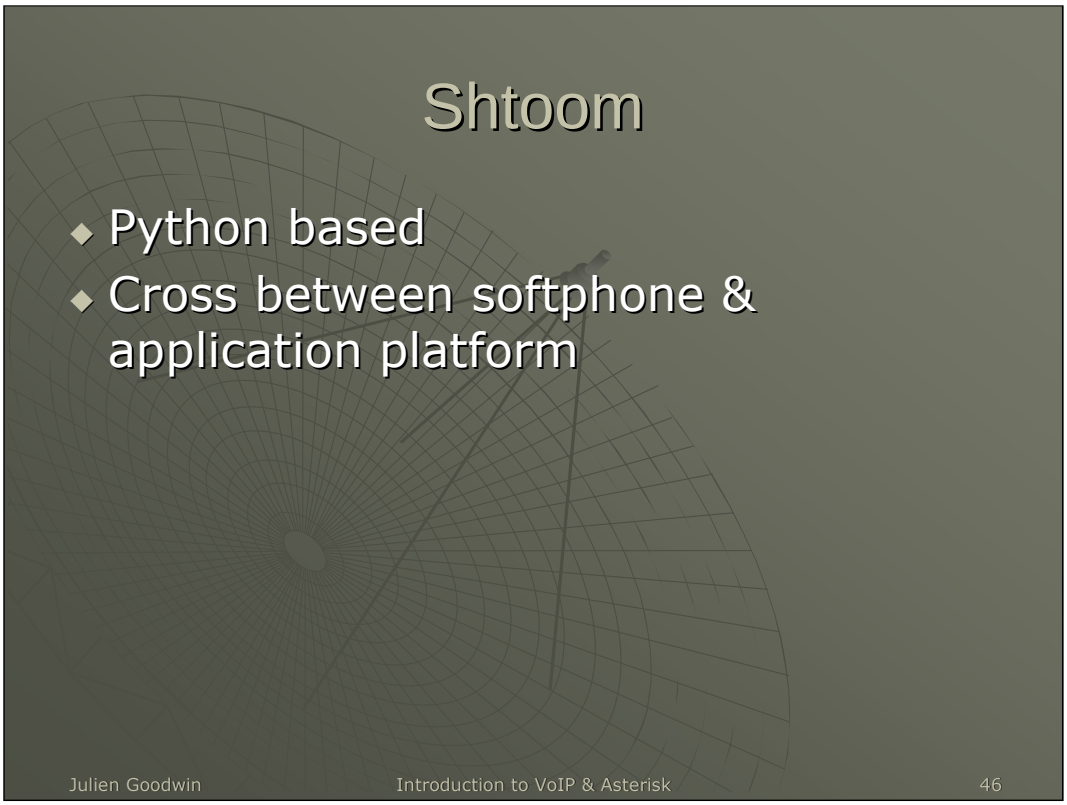
Phone running as software on the users PC

Problems:

Audio quality

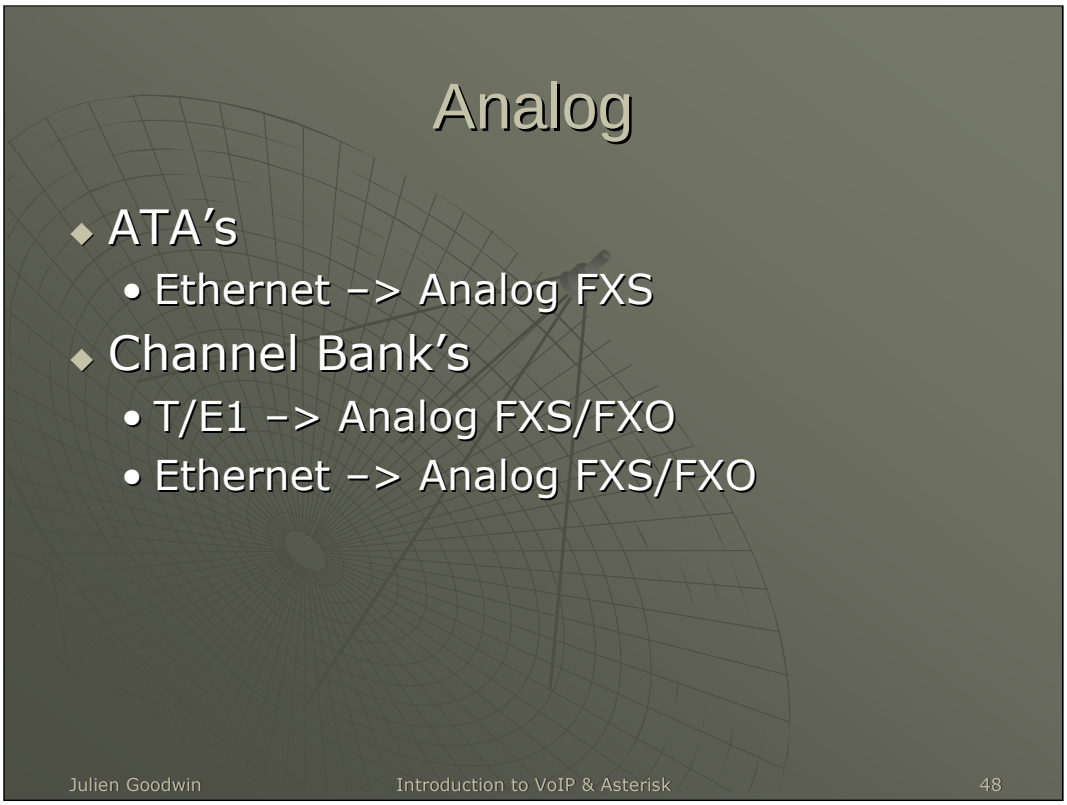
PC reliability

Bad usability



By Anthony Baxter – Python release manager

Non-VoIP Phones



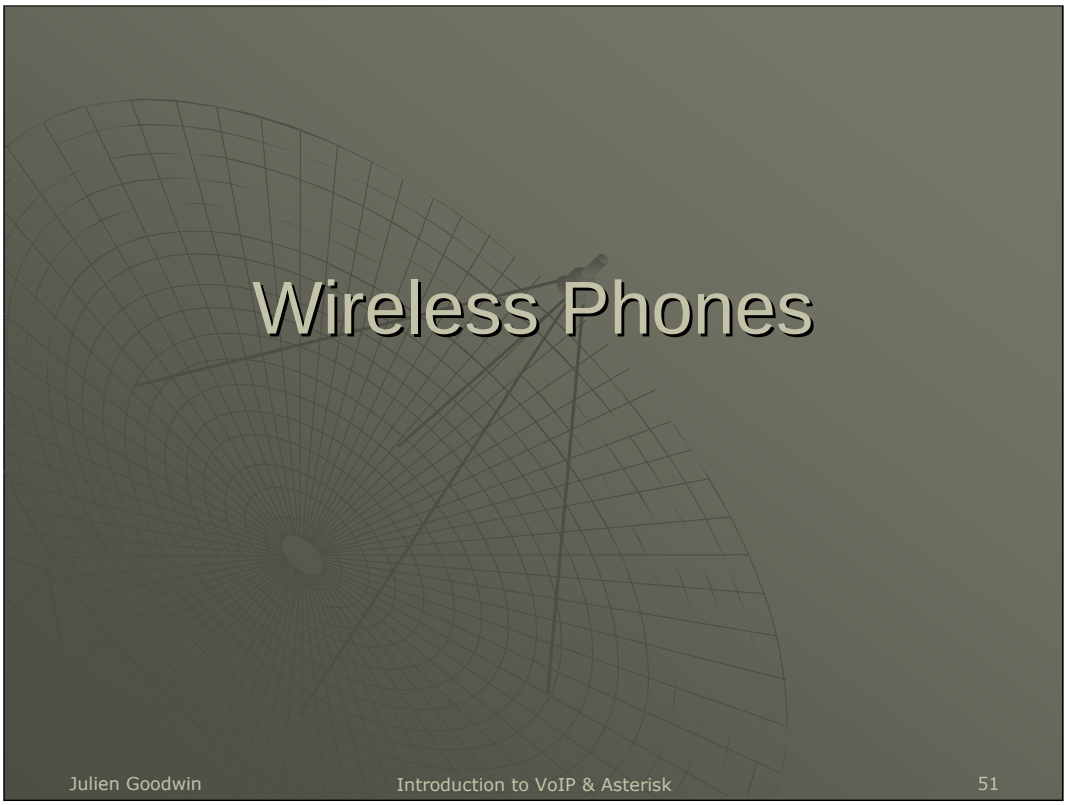
Use ordinary analogue phones, or cordless phones

ADSI

- Analog Display Services Interface
- PBX Features, on an analog link
- Graphical Menus
- Feature Buttons

Old PBX Phones

- Generally unusable
- Some gateway's exist
- Best to run services on * if possible
- Asterisk can configured to be transparent


Wireless Phones

802.11?

- Most suck
- Compromises
  - Battery Life
  - WiFi Power
  - Audio Quality
- Interference

Julien Goodwin Introducti

Interference – almost any additional traffic can cause issues for voip calls on wifi

There are some good ones starting to filter through, but a much better idea is…

Dect - Digital Enhanced Cordless Telephone

Kirk only supports H.323 and SCCP

Kirk ip500 is the multi base station equivalent of the ip600, uses ip600's as the base stations

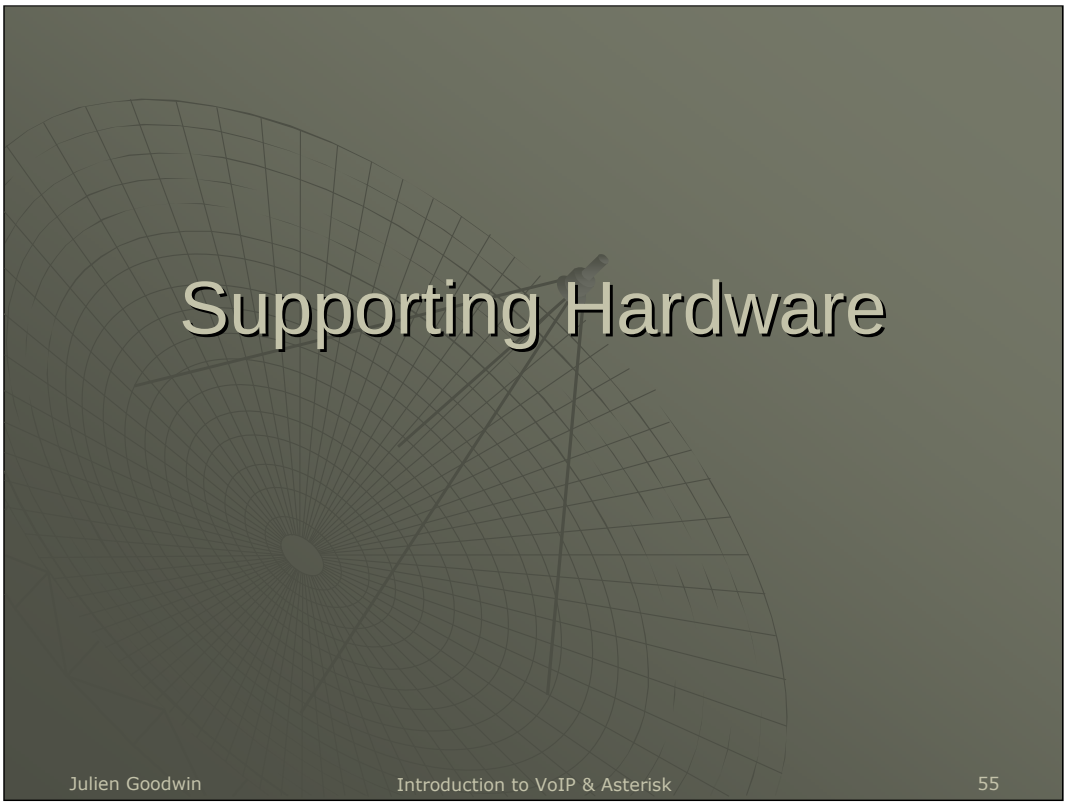
The 480i includes a dect base station as an option

Other

- Uniden UIP-1868

Julien Goodwin

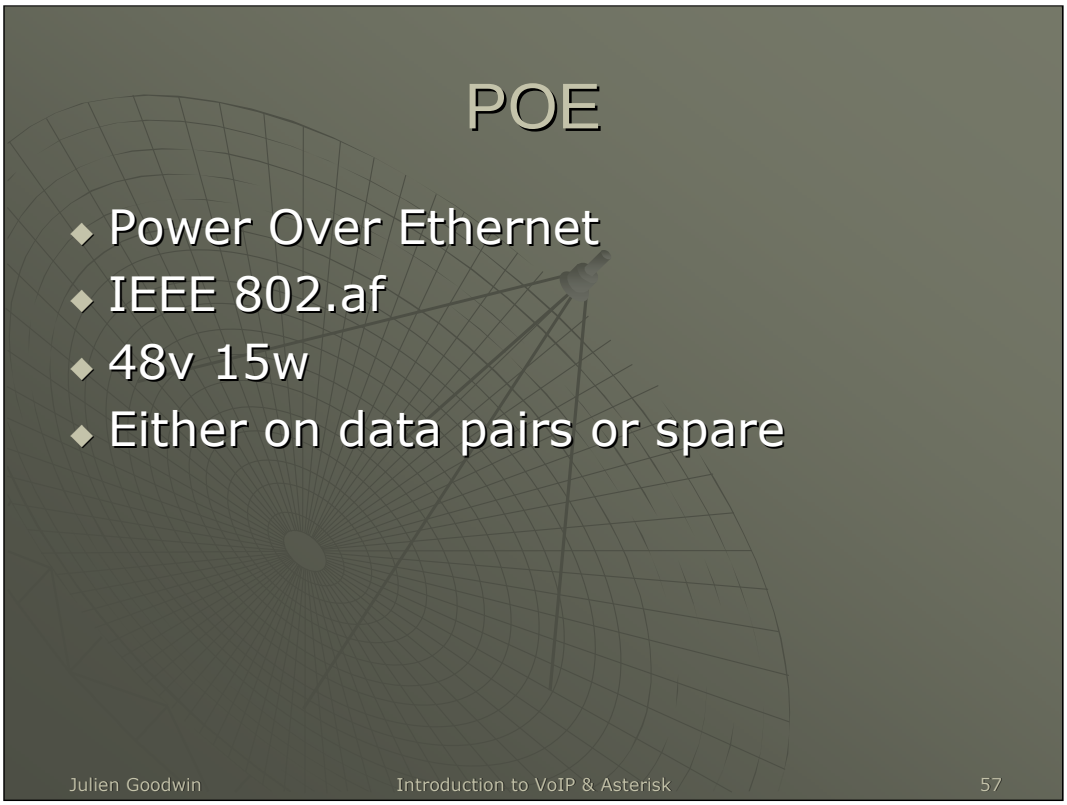
SIP phone


Supporting Hardware

T/E1 Hardware

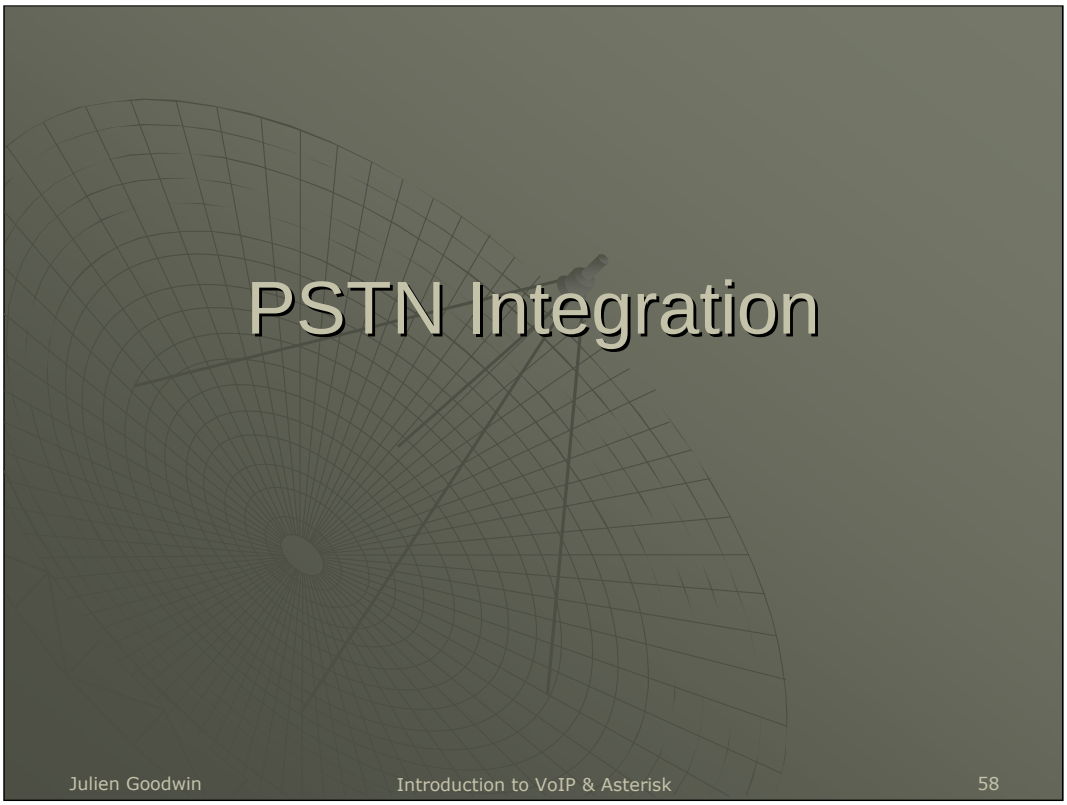
- Channel Banks
- Failover devices

The channel bank there is an ADIT 600, one of the most common channel banks in use today



Note that a pretty much all of the Cisco stuff produced before mid-2005 is non-standard

There was a demo of the 3com wall plate switch



Talking here about hardware that supports asterisk

Analog

- Internal:
  - Digium TDM400 – 4 port FXS/FXO
  - Digium TDM2400 – 24 port FXS/FXO
  - Digium TE* - 1/2/4 port T/E1
    - Use with a channel bank
- External
  - Sipura SPA-3000 – 1 FXS / 1 FXO
  - Larger Ethernet channel banks

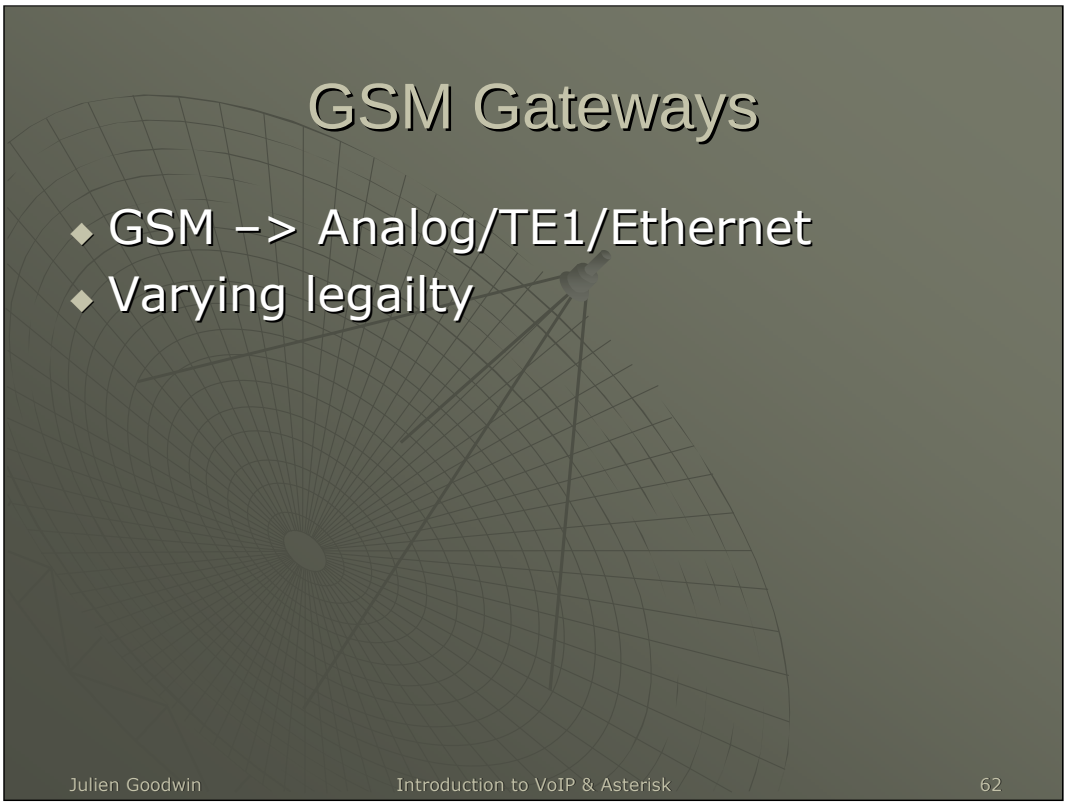
A company called sangnoma also produces TE1 cards, and is about to introduce a card that supports analog

ISDN

- Single BRI:
  - AVM Fritz
- PRI – OnRamp 10/20/30
  - Digium TE* cards
  - Eicon Diva cards

TDM – T/E 1

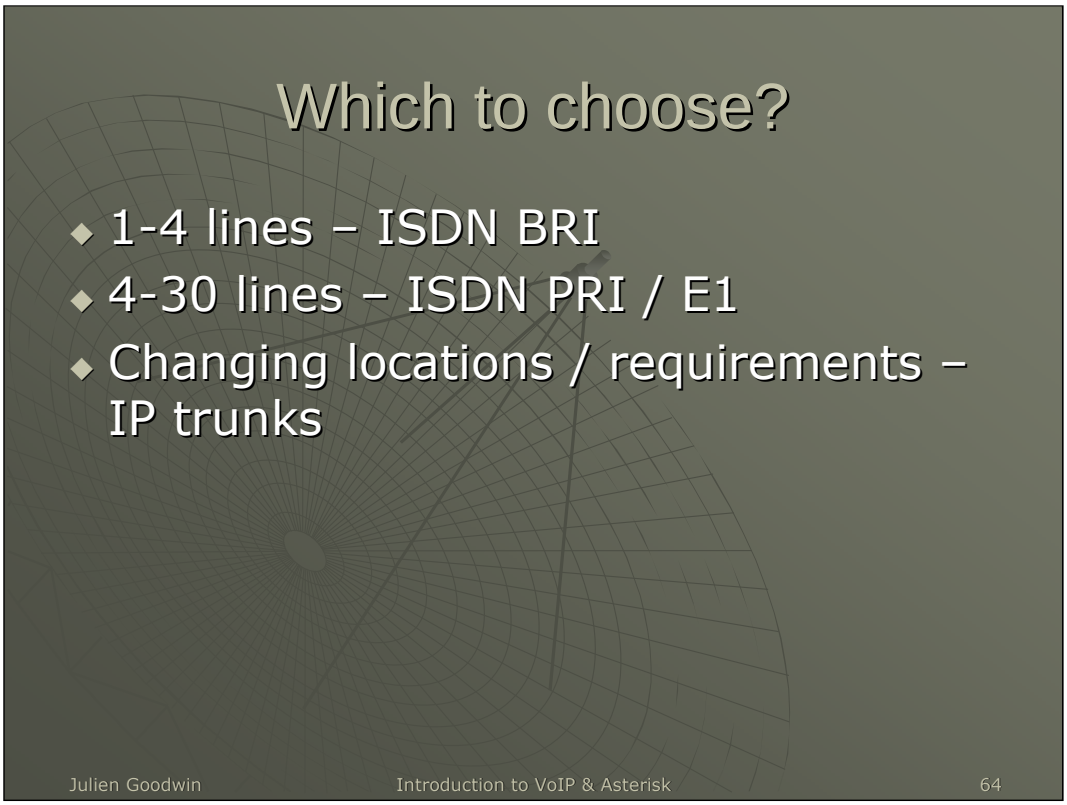
- Digium TE* - 1/2/4 port T/E1
- Sangnoma cards
- Eicon Diva cards



Try to limit use of gsm gateways to internal only (possible legal issues in some countries)

IP Trunks

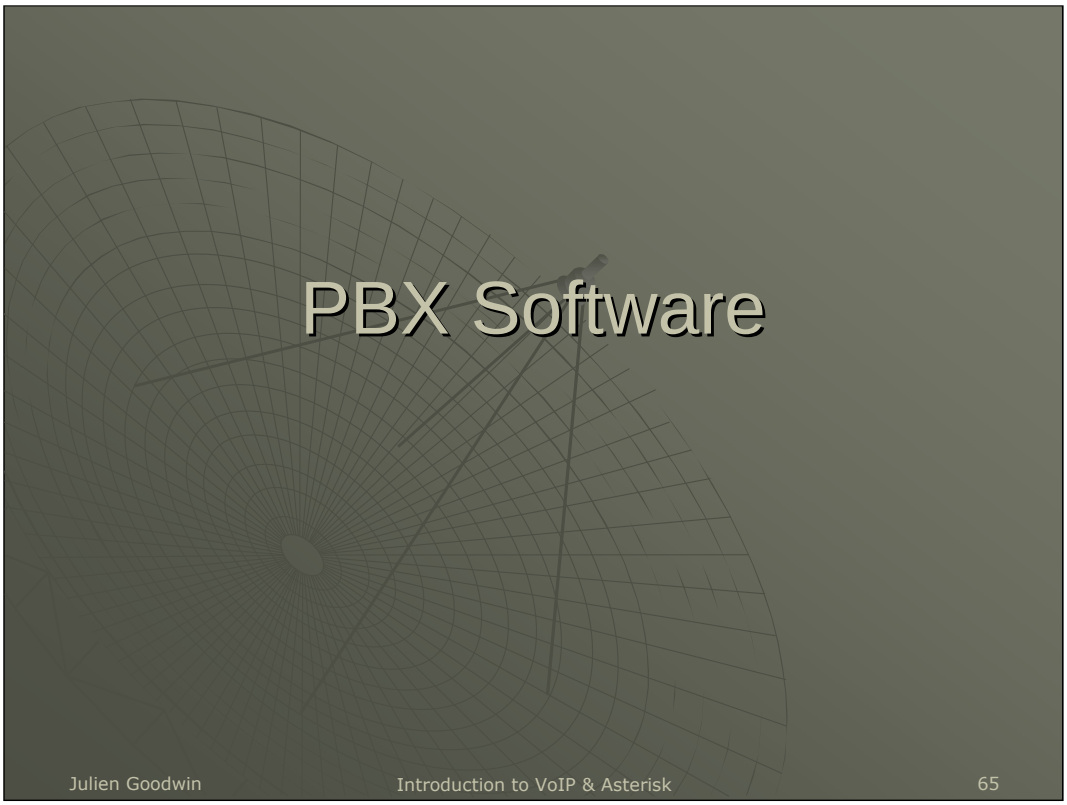
- Delivered over H.323 / SIP / IAX
  - Use a dedicated Ethernet if possible


Which to choose?
1-4 lines – ISDN BRI
4-30 lines – ISDN PRI / E1
Changing locations / requirements – IP trunks
Julien Goodwin
Introduction to VoIP & Asterisk
64

Avoid analog

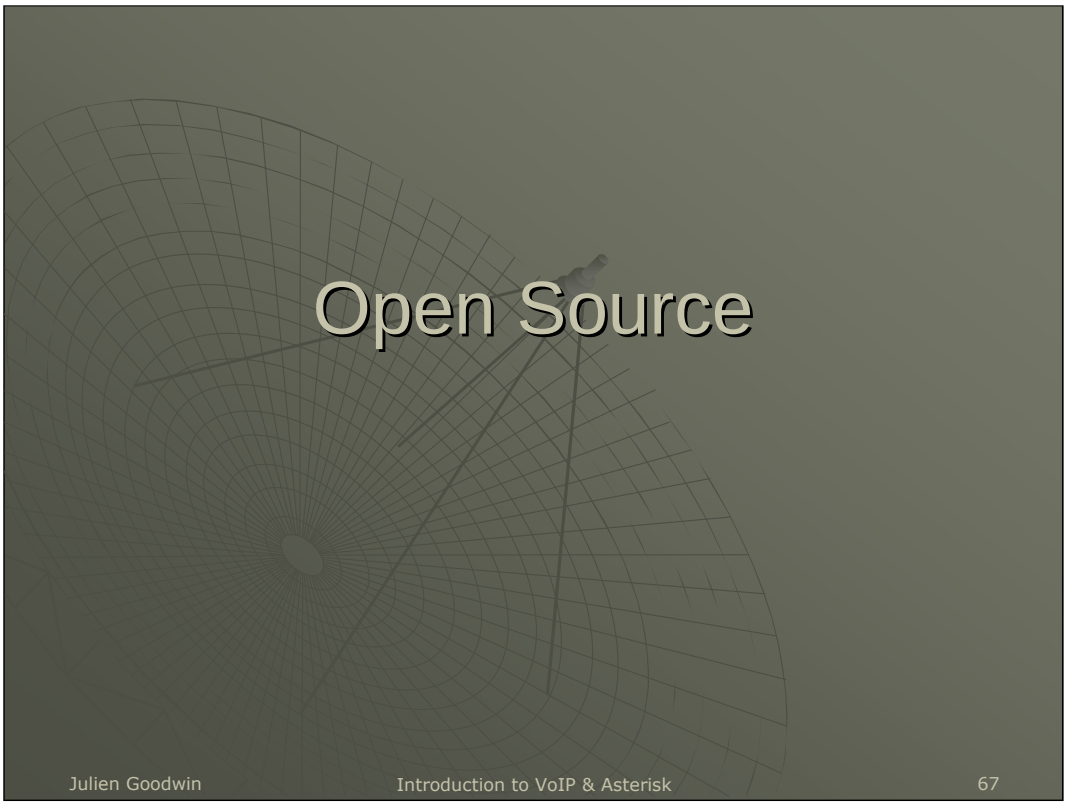
IP trunks also good when you have a large excess of bandwidth on an existing link to a voip provider

Obviously what you already have running will determine this a lot


PBX Software

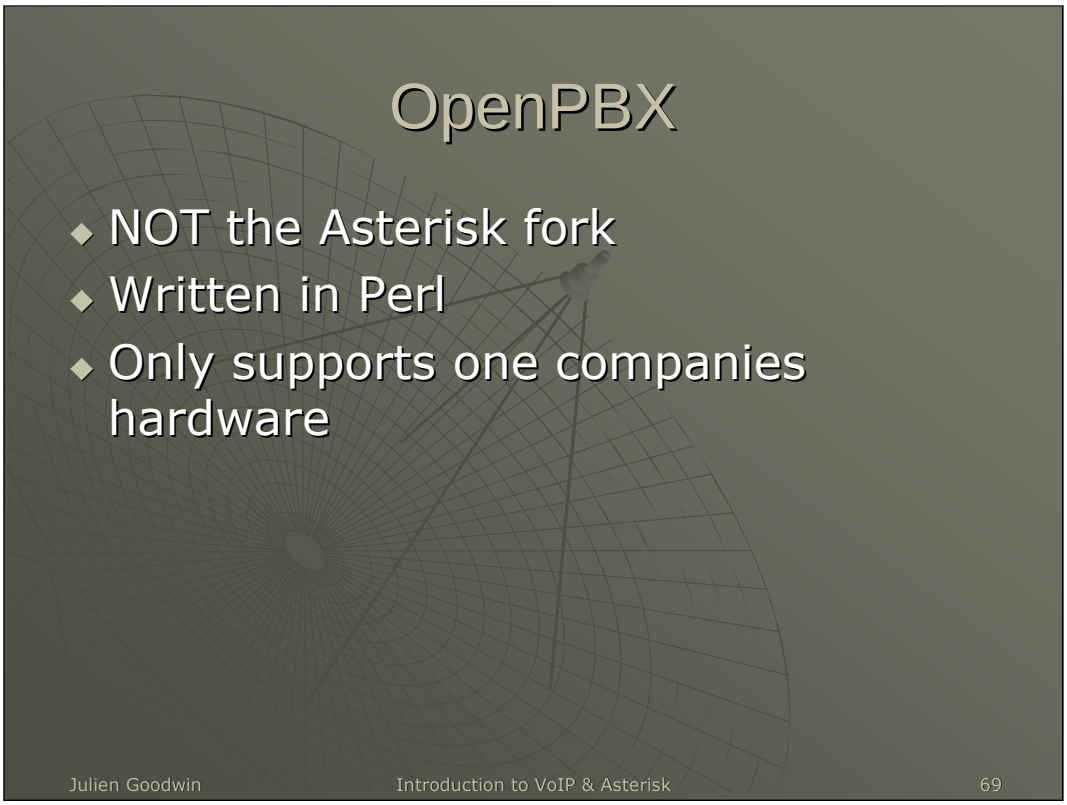
Commercial

- Cisco CallManager
- Avaya
- Broadvoice

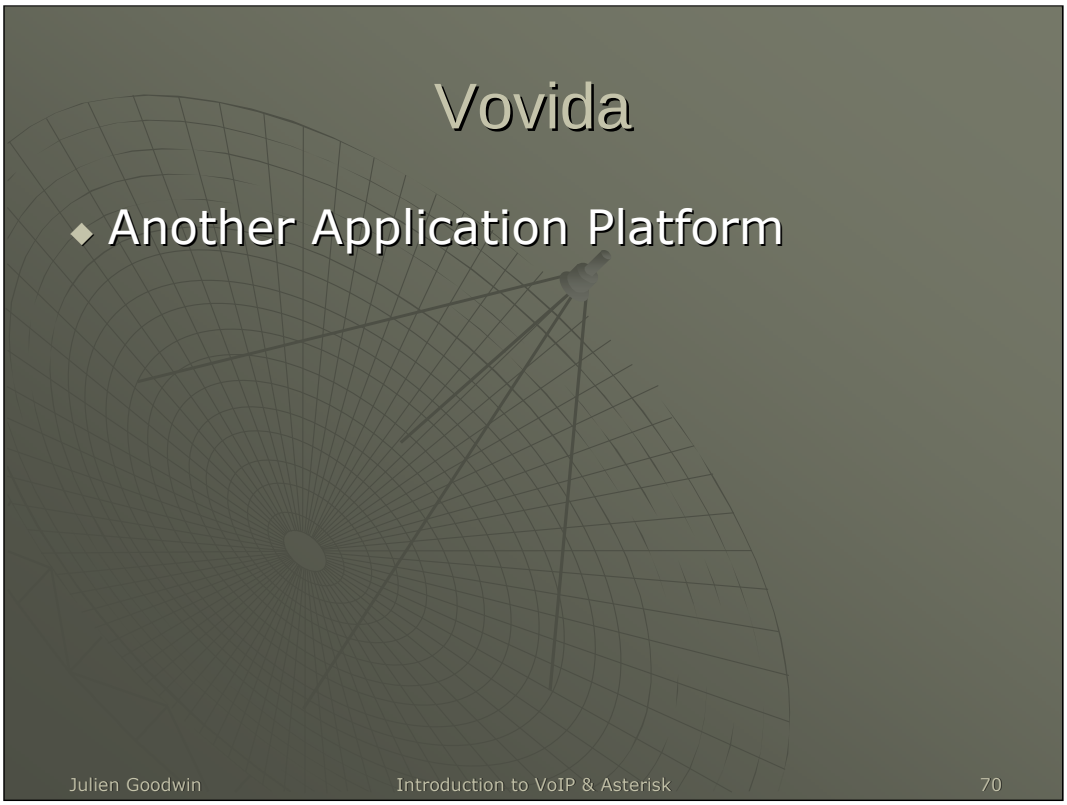

Open Source

GNU Bayonne

- More an application platform
- Pretty much dead



Small enough that you can understand the code fairly easily



Vovida comes from VOice VIdeo DAta. It is an site dedicated to open source Voice over IP software. Currently there is over 40 people working full time work and many more doing part time work on software for this site making it a fairly major project. People have been working on it since early 1999. It has software for SIP, MGCP, RTP, TRIP, Radius protocols, and a SIP based soft phone for Linux. In the future we are considering VXML, RTSP, SIP Proxy Servers, H.323 applications, XML provision systems, voice mail system, and whatever else there seems to be a strong need for.

Don’t know the status, web site is down

PBX4Linux

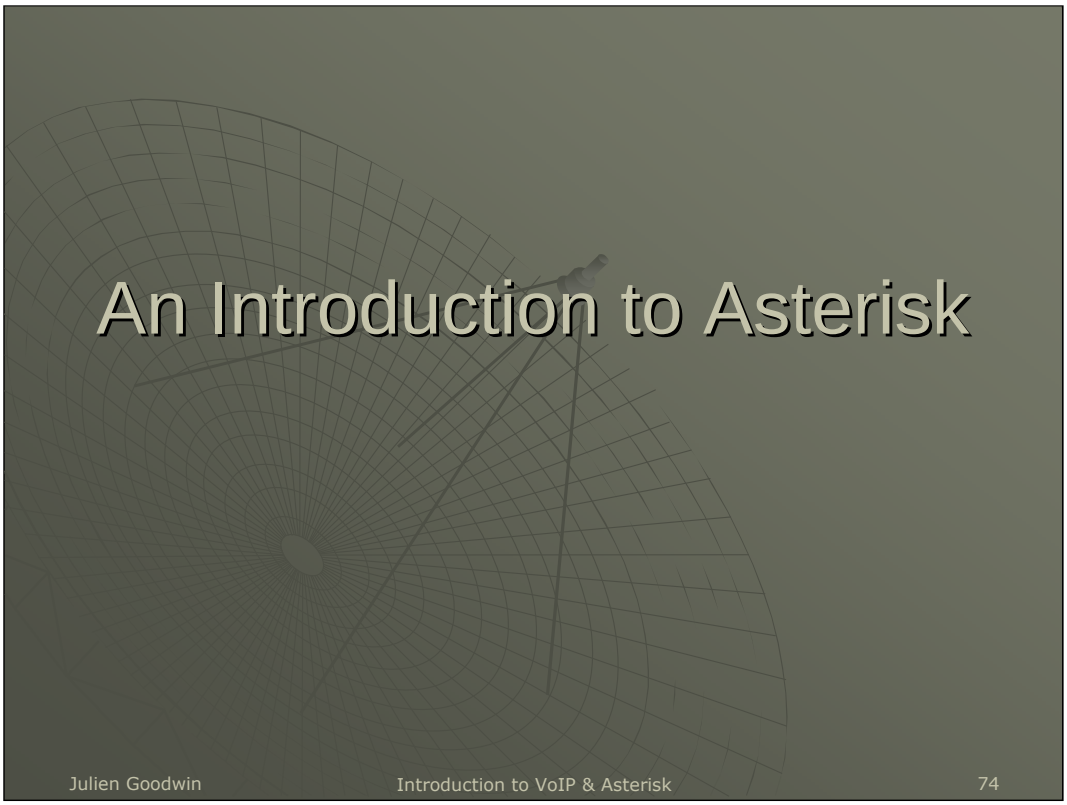
- A simple PBX
- Supports mISDN hardware only
- Very minimal VoIP support

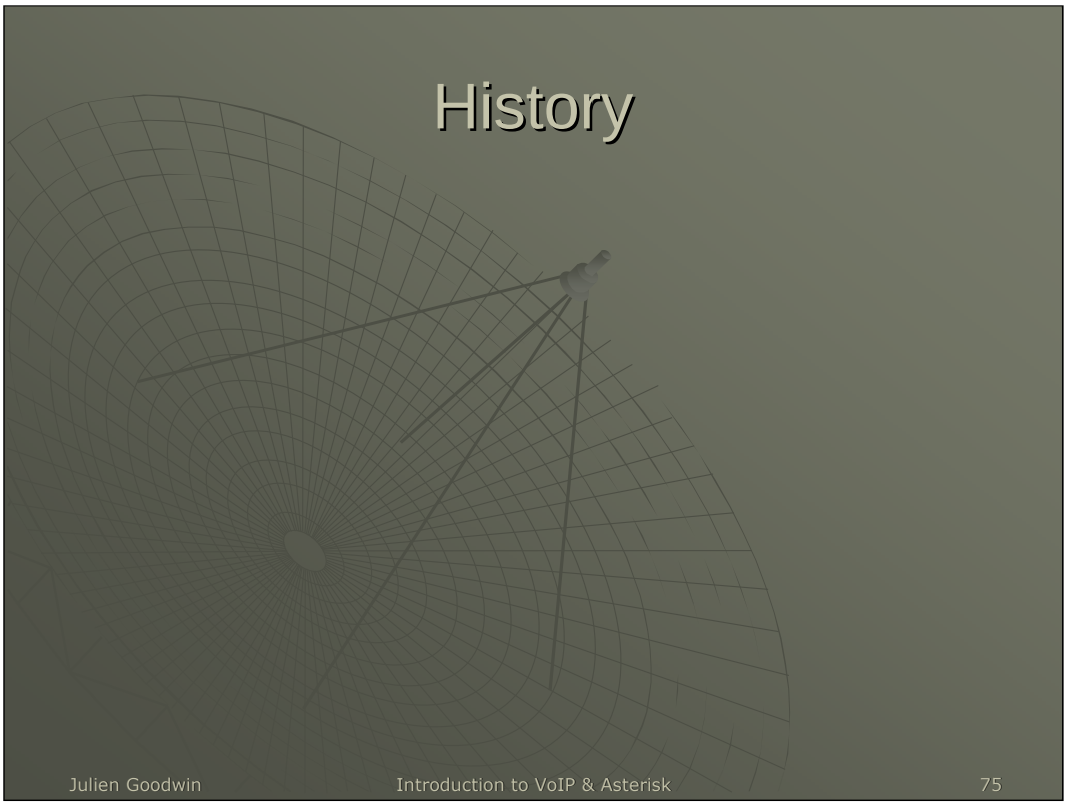
Asterisk

- The focus of today's talk
- More info later

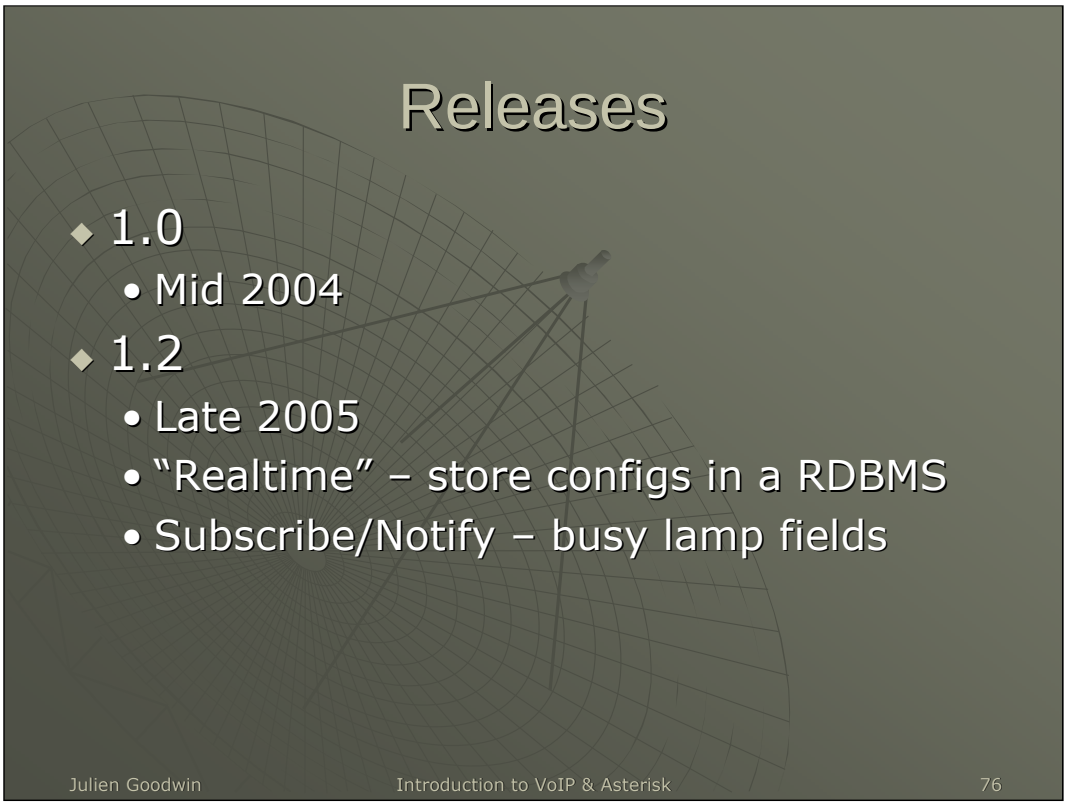
OpenPBX

- The Asterisk fork
- Little Benefit
- Less Support
- Not Recommended for new users


An Introduction to Asterisk



Slide missing?

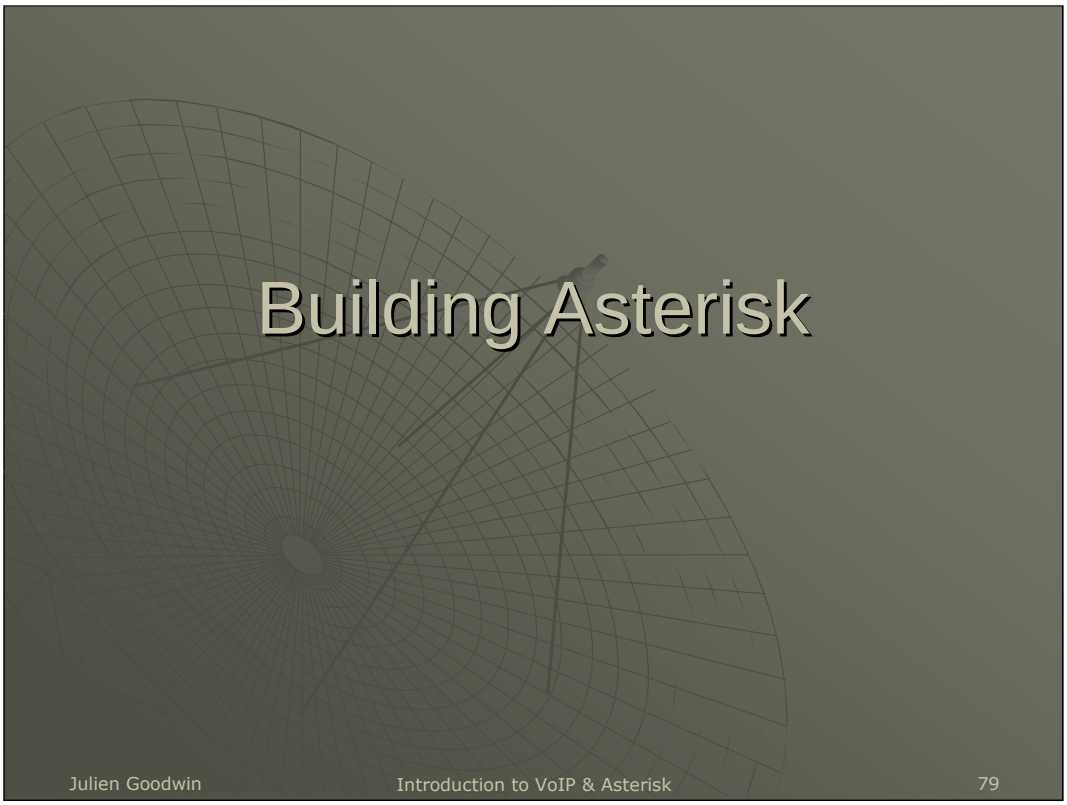


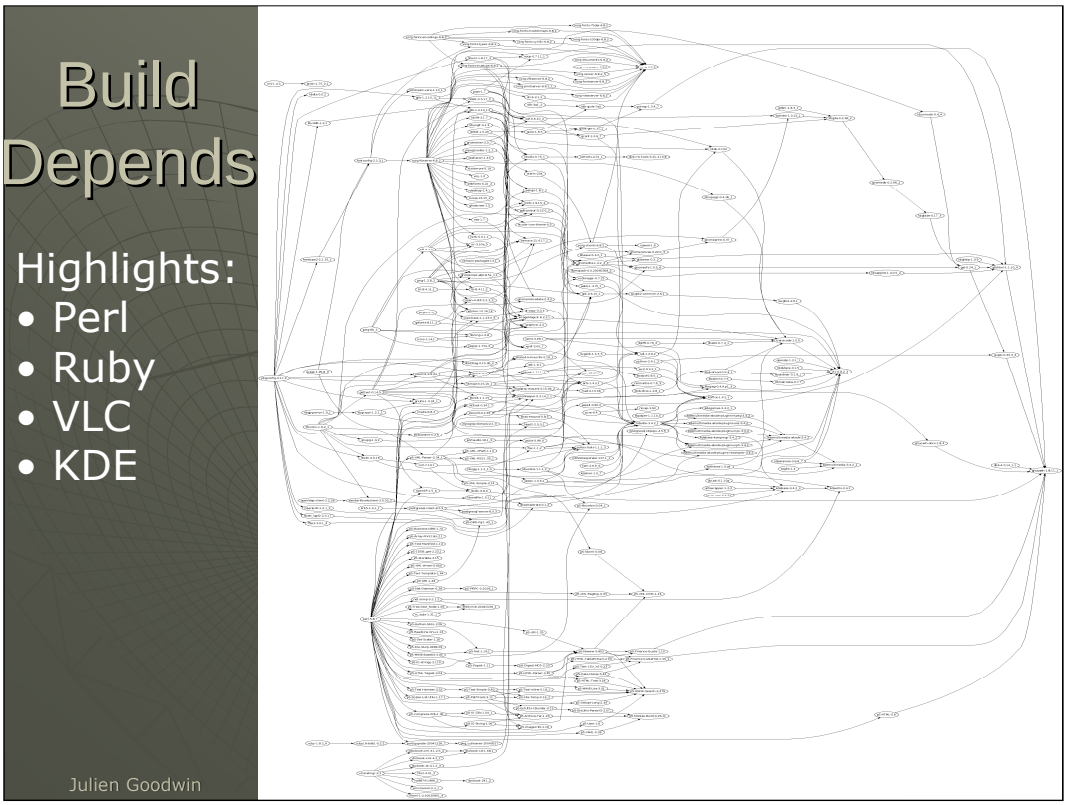
Realtime – live updates

Features

- Wide Protocol Support
  - SIP H.323 MGCP IAX Skinny/SCCP
- Several Programming Options
  - Management Console
  - Out of band applications
  - In band applications
- Ships with extensive suite of apps

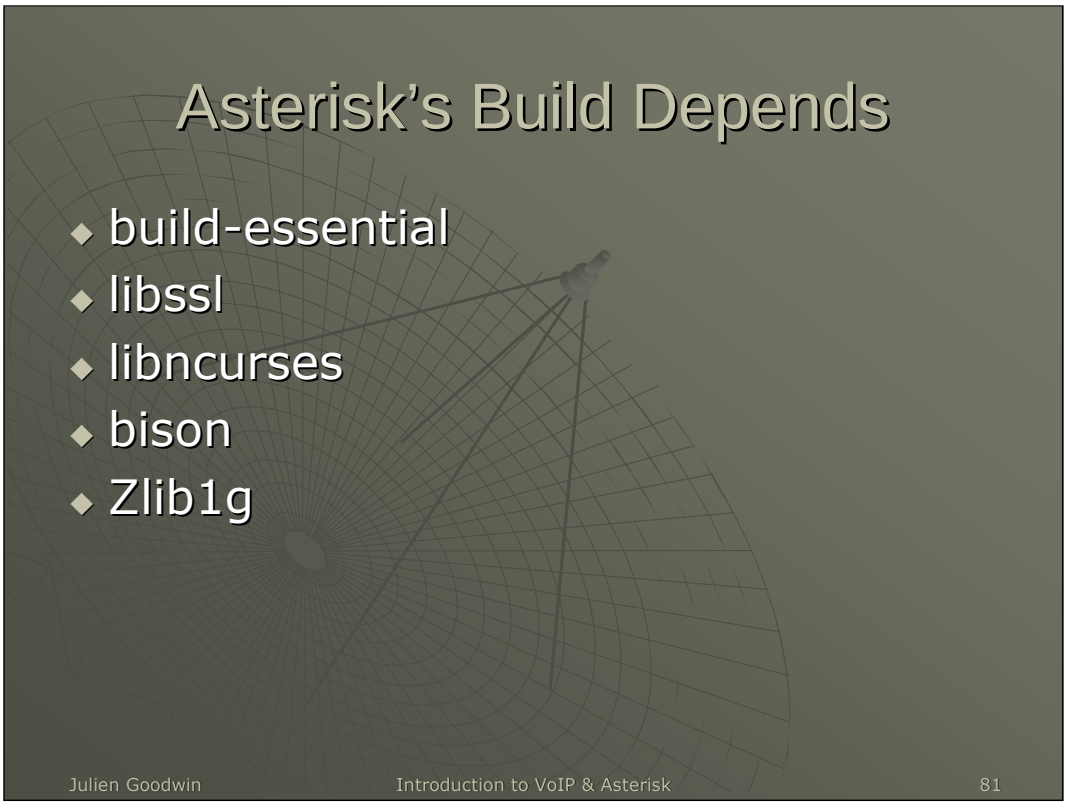
Building a SoHo PBX with Asterisk


Building Asterisk



Kdelibs, but still

But, actually that’s gnucash



For non debian people build essential pretty much equals:

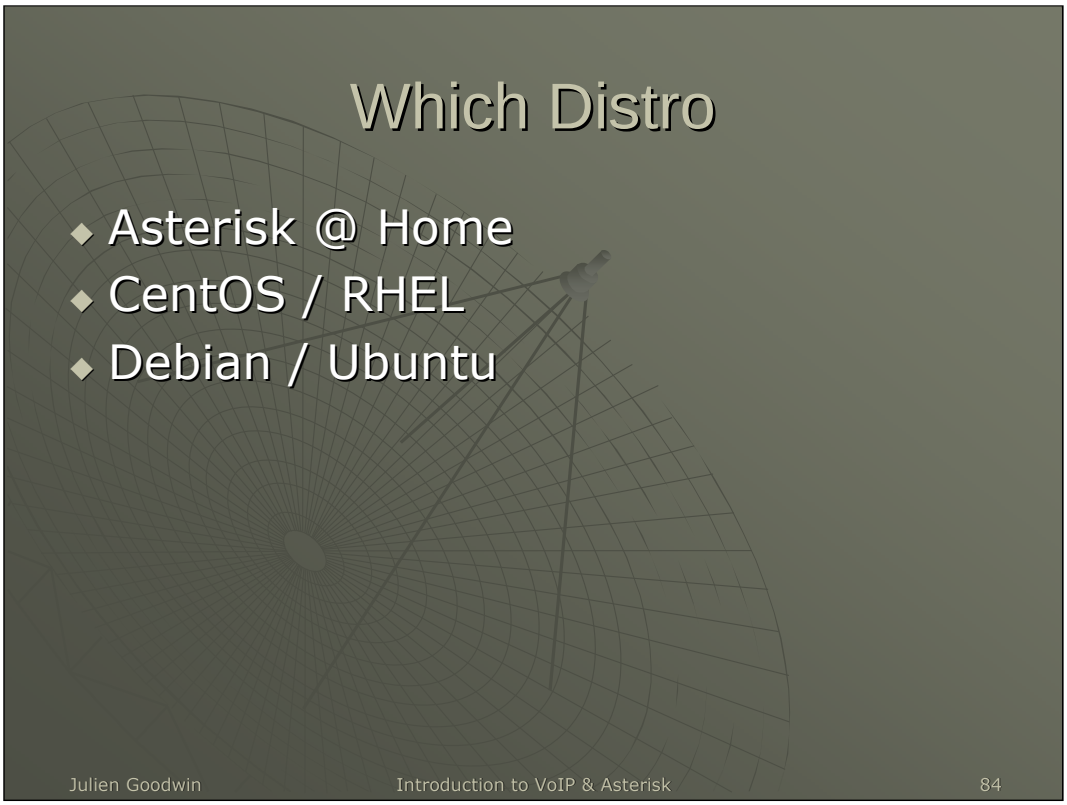
* GCC, G++, CPP, MAKE, packaging tools

Sizing your system – 1

- How many users?
- What codecs?
- What Features?
  - Voicemail
  - Voice Menus
  - TTS
  - AGI

Sizing your system – 2

- CPU Speed
- RAM
- Drives
- Motherboard
- NIC's



A@H is centos

Digium develop on rhel

Just a few things I'd like to cover, and then we'll switch over

Zaptel Drivers & Timing

- Drivers for Digium hardware called "Zaptel"
- Kernel modules, won't get merged
- Also used as a high quality timing interface for:
  - Music on Hold
  - Conferencing

1k interrups/second/card

Echo Cancellation

- Single wire analog –> Echo
- Echo cancellation fixes this
- Zaptel supports, some hardware may not
- New Digium hardware accelerated

Hold Music Selection

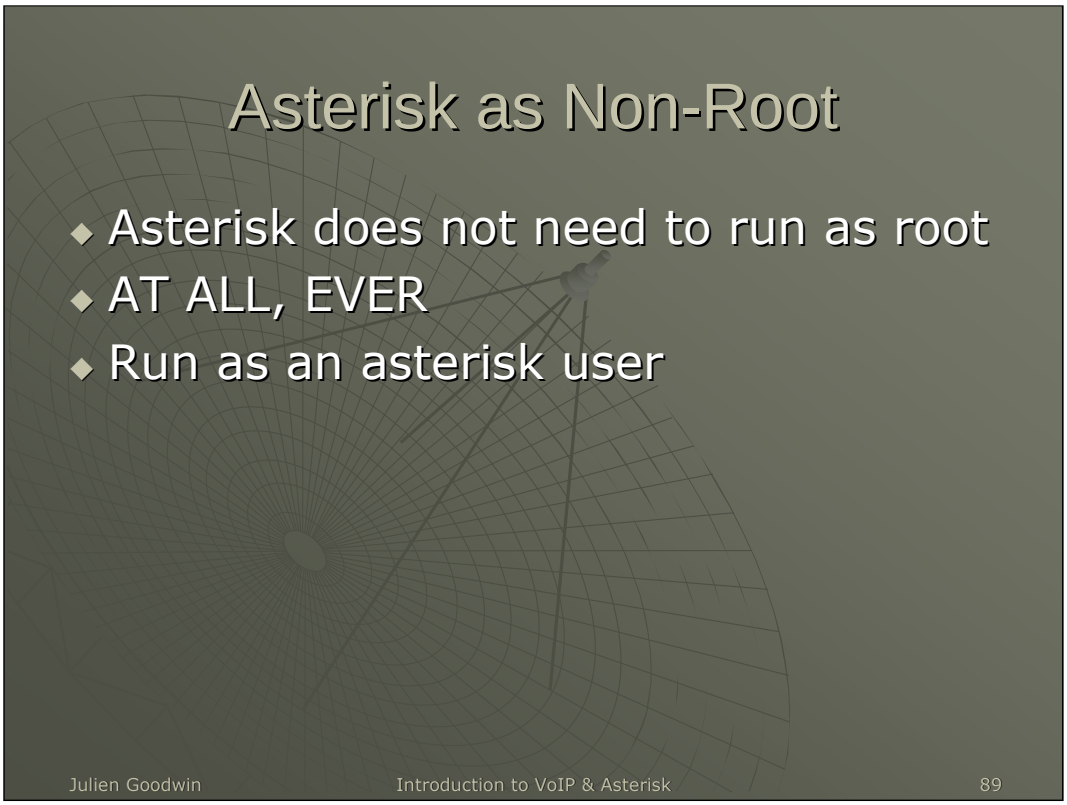
- This is what I normally give people
- This is what I thought might be more appropriate
- (Then I realised that I'd get lynched)
- So how about this?

The demos were (in order):

•A selection from the OpenBSD release music collection

•RMS singing the free software sonf

•The LCA 2006 audio miniconf mix



At all, ever – is unless you use a realtime process priority, which is very potentially error prone

Demo Process

- Installing from source
- Configure phone #1 (outgoing)
- Configure phone #2 (outgoing)
- Configure phones calling each other
- Configure trunk extension (outgoing)
- Configure trunk extension (incoming)
- Fix any potential security issues
- Add additional features / phones

Follow this process when installing your own pbx